SOUND ON SOUND ™

MASCHINE+

NI free the Maschine from the computer!

Bob Clearmountain:
Mixing Springsteen
"I don't understand how anyone gets a decent mix in any DAW."

Steinberg Cubase 11
The update you've been waiting for?

New Year, New Studio
21 upgrades to make for 2021

WIN
AUSTRIAN AUDIO
OC818 DUAL SET PLUS
WORTH $2199

REVIEWS: SPITFIRE / TOONTRACK / FOCUSRITE / KORG / ZOOM / OCEAN WAY / ACCESS ANALOG / NUGEN / ARTURIA / LEAPWING

TECHNIQUE: LCR PANNING / DAW WORKSHOPS / WHAT NOT TO WORRY ABOUT!

THE LUXURY OF CHOICE

Back in the '80s, tastes in music were narrow. We were passionate about music, but we were wedded to our chosen scenes. The idea that a thrash metal fan might also enjoy the Smiths was utterly alien. The only thing that bound us all together was a shared revolt against the music of the '70s!

The Internet seems to have swept all that away. Today's teenagers see nothing odd about playlisting the Beatles and Sinatra cheek by jowl with trap-metal and drill. Music of the past is seen as a canon to be added to, not an embarrassment to be left behind. This seems to me a very positive change, and I wonder if it's being reflected in the ways people make music.

Back in the '70s, liner notes famously announced that no synthesizers were used in the making of Queen albums. What young band today would see that as something to boast about? Today's musicians are happy to use whatever will get the job done in the quickest and most effective way. No-one thinks it's somehow cheating to use the 'wrong' piece of kit to create a sound or express a musical idea. And whilst I harbour reservations about AI-based tools for mixing and mastering, I suspect that's just me. I'm sure most people are just getting on with making music and using these tools on a 'whatever works' basis.

What makes this all possible is choice. And when it comes to music production, the most fundamental choice is whether or not to use a computer. That choice has been around for a long time, but for many years it was a pretty irrevocable decision. You either bought an Atari and learned MIDI sequencing that way, or you went down the hardware route with grid-based grooveboxes. The path you chose profoundly shaped the music that came out at the end.

That chasm was then bridged by products like Ableton's Push, Akai's MPC Live and the original NI Maschine, which successfully combined the hands-on charms of grid-based sequencing with the open-ended power of mainstream computing. But in doing so, they lost the self-contained, go-anywhere freedom that characterised the original MPC and related designs. That freedom is now returning, and in the current generation of music technology, we may be seeing a reaction against the idea of using one super-powerful tool to do everything.

That's not to say that Akai's MPC Live, Elektron's Octatrack or NI's new Maschine+ aren't powerful. They are, hugely so. But the emphasis is no longer on power alone, or on limitless possibility. It's on immediacy and accessibility. Manufacturers are acknowledging that every tool has its own strengths, and that it's best to play to those strengths. We're no longer in a position where we have to commit to only one way of making music, or using a single tool to do everything. And whilst buying more kit can never replace good ideas, it has to be a positive that musicians can afford to own multiple tools and use each one where it seems most appropriate. Choice is, indeed, a luxury! ◢◢◢

Sam Inglis
Editor In Chief

> "We're no longer in a position where we have to commit to only one way of making music, or using a single tool to do everything."

No other pro audio retailer offers more for the recording industry than Sweetwater.

RODE RODECaster Pro Podcast Production Studio

NEW!

Dynaudio Core 59 3-way Powered Studio Monitor – Black

PreSonus ATOM SQ Keyboard/Pad Hybrid MIDI Keyboard/Pad Performance and Production Controller

Experience the Sweetwater Difference Today

Best Prices
thanks to our huge buying power

Massive Gear Selection
of nearly 40,000 products from today's best brands

Expert Advice
and attention to detail from our highly knowledgeable Sales Engineers

Free Technical Support
from trained professionals on every product we sell

Free 2-year Warranty
and factory-trained service technicians to solve any issue

Fast, Free Shipping
from our centrally located warehouse in Fort Wayne, IN

Shure SM7B
Cardioid Dynamic
Vocal Microphone

Universal Audio Apollo Twin X DUO
10x6 Thunderbolt Audio
Interface with UAD DSP

Mojave Audio MA-200
Large-diaphragm Condenser
Microphone – Satin Nickel

NEW!

Warm Audio WA87 R2
Large-diaphragm Condenser
Microphone – Nickel

NEW!

prophet-5

Sequential Prophet-5
61-key Analog Synthesizer

Sweetwater

Music Instruments & Pro Audio

(800) 222-4700   Sweetwater.com

IN THIS ISSUE

January 2021 / issue 3 / **volume 36**

FEATURES

100 CUBASE 11

ON TEST

WORKSHOPS

MATT HOUGHTON

Korneff Audio
Pawn Shop Comp 2.0

Compressor Plug-in

We check out Korneff's vibey take on a tube-laden FET compressor.

Back in *SOS* August 2020 I reviewed Korneff's Talkback Limiter (www.soundonsound.com/reviews/korneff-audio-talkback-limiter), which was a wonderfully warm and teasingly tweakable model of SSL's Listen Mic Compressor. Korneff have taken a similar 'tweaker's delight' approach to their new Pawn Shop Comp 2.0, and while this a very different compressor — this time it's a virtual FET compressor with tube amplification stages — the interface will feel familiar to anyone who has tried the Talkback Limiter.

You're first presented with a GUI (below) that offers only the few controls you'd usually need to get the compressor doing its thing. Namely, there are Attack, Release, Threshold and Ratio knobs. There's also an on/off button (to switch the compression in and out, but with the signal always passing through the virtual analogue circuitry) and an Auto make-up gain button. A gain-reduction meter tells you what's going on.

That the initial preset had the threshold pegged anti-clockwise for maximum compression was indication enough that this device is intended as much for audio abuse as control, but it's obvious to your ears too when you run pretty much any source through at the default settings. That said, while this thing will never sound 'clean' exactly, there's plenty of control on offer, and you can dial in much more subtlety if required.

But I've so far only told you the boring half of the story. As with the Talkback Limiter, click on the Korneff logo and you're teleported to a tweaker's paradise: a skeuomorphic second 'page' loosely resembling the innards of a hardware compressor. Here, you have input and output trim knobs, oversampling selectors, a wet/dry mixer, and a range of parameters to fiddle with in search of different sonic characters. You can also set the GUI to a range of sizes up to 200 percent (which was big enough to spill outside my 2018 MacBook Pro Retina screen) so hi-DPI fans should be happy.

At the top, the virtual preamp valves can be swapped for other types, with 12AX7, ECC83 and 5751 models to choose from. Three output transformers have been modelled too, the options being Nickel, Iron and Steel. The lower half presents Preamp (gain) and Bias knobs, Weight (63 or 171 Hz) and Focus (1.6 and 2.4 kHz) controls and Operating Level pot and a wet/dry blend knob. You can also opt for different types of resistor and transistor. The manual does a good job of explaining

what these do on a technical level, but the whole point here is that the user doesn't really need to understand all of that: there's always pleasing character on offer and, once you've chosen suitable compression settings for the source, your job is simply to tweak the second-screen knobs and switches, select substitute components, and keep listening until your ears are greeted with optimum mojo. It never takes long to find something cool, and while the knobs always change the sound more than swapping components does, the latter is nonetheless always interesting.

So on what should you use this compressor? Well, anything that demands to be dirtied up or is crying out for more grit and attitude. It adds a lovely warm-sounding distortion that I thought perfect for rock and hip-hop vocals, and for electric and bass guitar more generally. There's plenty of scope to make drums push, pull and crunch, while subtler settings might offer interest on a group or master bus. But for me, it really shines as a parallel vocal 'crusher', cranking up the attitude and helping a part cut through a dense mix. With a slowish attack, medium-slow release, and highish ratio (by normal standards; this one can go up to 50:1!), 9 or 10 dB of gain reduction will get you in the ballpark. Then use the Bias, Weight and Focus controls to do the heavy lifting on the second page and back off the wet/dry mix to taste. Recommended. ◢◢◢

summary

A nice, vibey compressor plug-in that should please those with a penchant for tonal tweakery!

$ $99.99.
W https://korneffaudio.com

JOHN WALDEN

Spitfire Audio
BBC Symphony Orchestra Discover

Sample Library

Dave Stewart was hugely impressed when reviewing Spitfire Audio's BBC Symphony Orchestra sample library in the November 2019 issue of *SOS*. While BBC SO offered excellent value, its price made it a substantial investment so, in what looks like a sensible — and shrewd — move, Spitfire have now rebranded the original library as the 'Pro' version and introduced two additional tiers, Core and Discover, with more streamlined feature sets and more accessible prices.

Core is still superbly specified, but the subject of this review is the entry-level Discover edition and, while there is undoubtedly some very significant streamlining going on, it still provides what is essentially a full suite of orchestral instruments covering all orchestral sections. However, what's perhaps most remarkable about BBC Symphony Orchestra Discover is that (a) it is potentially available for free (or $49 depending upon your level of patience) and (b) the entire sample library occupies only 200MB (no, that's not a typo). How much orchestra can you actually get into a free library of 200MB? It's time to (BBC SO) Discover…

What You Get & What You Don't

Crammed into that miniscule 200MB are 33 instruments and a total of 47 articulations (playing techniques). In contrast, the numbers are 23GB and 600GB, 42 instruments and 46 instruments, and 305 and 435 articulations respectively for the Core and Pro editions. What you do get in Discover though is a broad palette of instruments. In the strings, this includes first violins, second violins, violas, celli and double-basses. For the woodwinds, we have piccolo,

Want a full orchestral sample library for free? Spitfire have got you covered…

flutes, oboes, clarinets and bassoons. The brass section includes horns, trumpets, tenor trombones, bass trombones and tuba. Finally, the percussion section provides harp, celeste, timpani, marimba, xylophone, glockenspiel, tubular bells and a range of untuned percussion. In terms of the articulations, all instruments include long notes (sustained with a touch of vibrato) and short notes (spiccato/staccatissimo), while

the strings also include pizzicato (plucked) and tremolo (with the exception of the double-bass).

In terms of what you don't get, in addition to fewer instruments and fewer articulations, there are no round-robin samples and only a single dynamic layer; volume (via velocity or mod wheel) has to be used to provide the bulk of the playing dynamics for individual instruments. Discover also excludes true legato samples and solo players. The other limitation compared to Pro is that you only get a single mic position, although this has been very well chosen (not too ambient) and there is always reverb if you want more room.

Minimal UI

Discover's UI is very straightforward. You get an info line across the top, while the upper half of the display provides you with two routes to select a patch; either via the drop-down browser or simply by clicking on

»
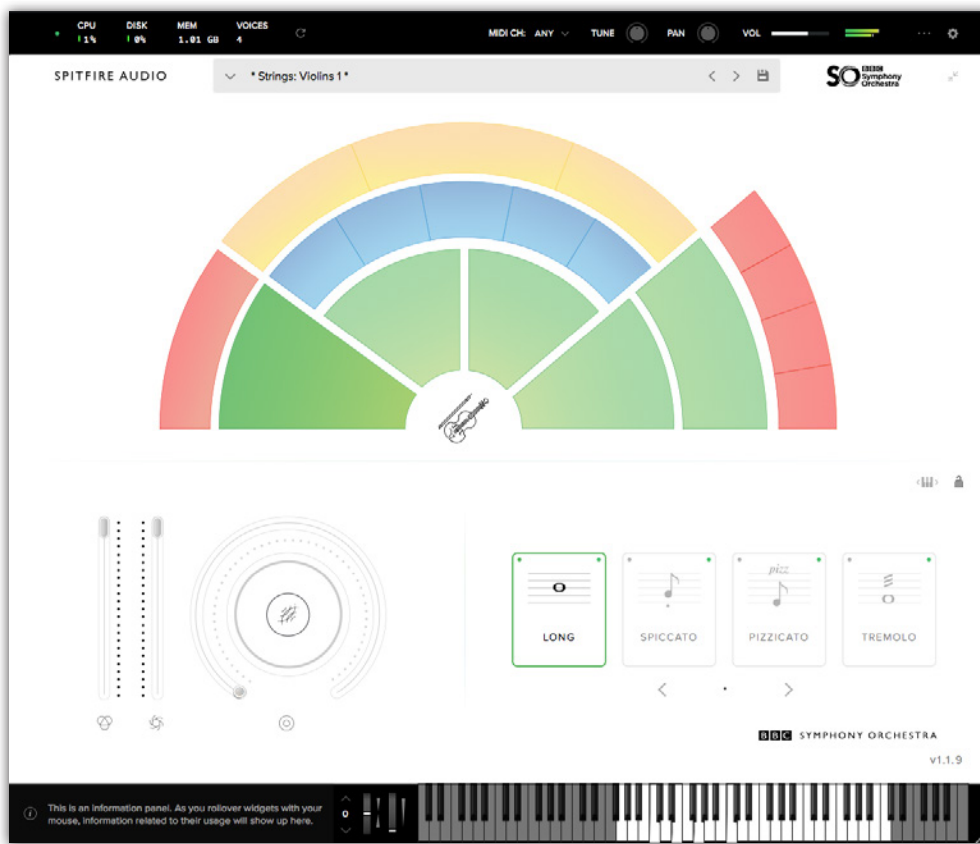
THREE
EDEN
TUBE
EDEN
MICROPHONES
EDEN
IN ONE

LAUTEN AUDIO

LT-386

DESIGNED AND BUILT BY
THE LAUTEN AUDIO FAMILY IN SILICON VALLEY, CA

» the colourful orchestral floorplan. The lower half of the display provides controls to adjust volume and the level of the internal reverb. Buttons are provided to switch between the available playing articulations and keyswitches are also available starting around the C-2 region of the MIDI keyboard.

Discover's RAM requirements are obviously light and, in testing on my own (modest by today's standards) iMac, I was able to run a session with every articulation (47 instances of the plug-in) playing at the same time without my system making a murmur. The UI's CPU and disk meters are useful in this context, but I would imagine you could have this full orchestral setup running on a pretty basic host computer without too many issues.

It's also worth noting that Spitfire's website has pre-configured downloadable template files for each BBC SO edition available for most of the major DAWs. I downloaded the two available for Cubase and they are a useful starting point for new users.

Sound For Pound

While you obviously can't conjure the finesse from Discover's sounds that you would from Core or Pro (or any other more deeply sampled orchestral library), in use, Discover manages to sound way better than a 200MB footprint suggests it has any right to. It obviously helps that the underlying samples are very good indeed. The combination of long and shorts notes for every string, brass and woodwind instrument is both an obvious and sensible choice and it allows you to write slow sustained parts or more up-tempo melodic runs.

While all the sustained instruments must feature some sample looping, it would appear to have been beautifully done and the results are very smooth. All the short articulations have a nice crisp attack to them and, if you want to add a little extra edge to your sustained notes, the long and short notes can easily be layered if you run each articulation in a separate instance of the plug-in. The additional string articulations are also very welcome, and all the pizzicato instruments are very useable. You do have to make good use of the mod wheel and velocity to provide as much by way of dynamics as you can.

The only other comment I would make is that the string presets do include some noticeable low-end bow noise. I guess this adds to the realism but, if you want to keep the bottom-end of your mix tidy, a little low-cut EQ on the required tracks is an easy solution.

Moving On Up

As mentioned earlier, you can obtain Discover for free or you can pay $49. The free route requires you to complete a questionnaire for Spitfire and then wait approximately two weeks for your request to be dealt with. If you chose the paid route, Discover is available for immediate download and, as Spitfire Audio rather wonderfully pay royalties to their performing musicians, you can feel good that you are contributing to that process.

Discover is also a route to either Core or Pro and upgrade pricing means anything you have already paid is discounted on the price of the next tier you
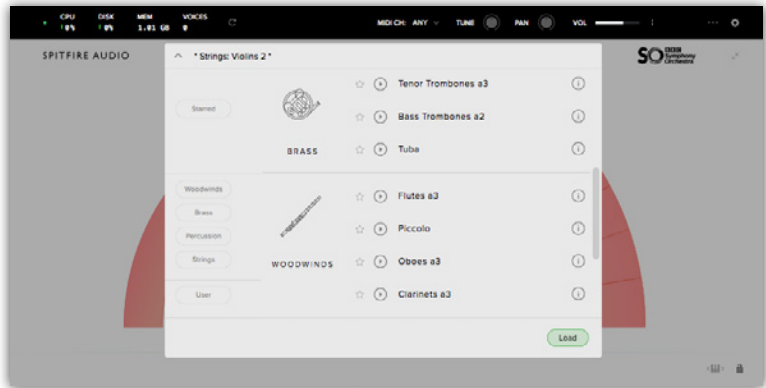
Despite the modest size, you get a broad suite of orchestral instruments to work with.
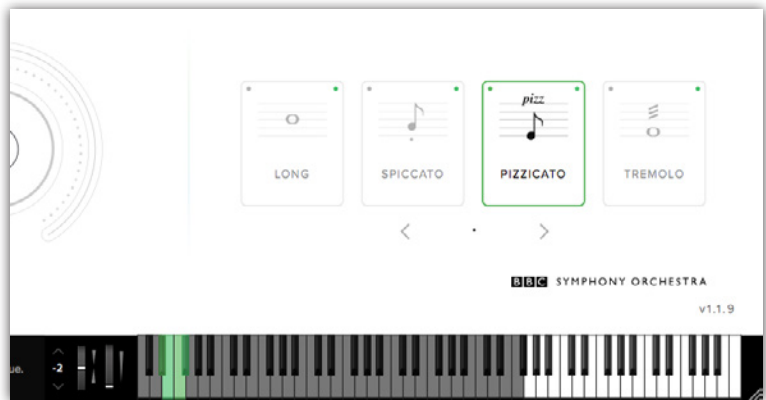
Articulations can be switched within the plug-in or using keyswitches (shown in green on the mini keyboard graphic).

wish to buy. One other clever technical feature in this regard is what Spitfire call Mode Switching. This allows projects created using one edition to be opened with another while the plug-in intelligently substitutes instruments depending upon which editions you are moving between. Without access to Core or Pro, I didn't get the opportunity to test this, but it's a cool feature.

Win, Win

Discover is undoubtedly a 'win' for Spitfire Audio on a number of levels, one of which is as an accessible entry point into the BBC SO ecosystem for budding orchestral composers. It's also a 'win' for potential users. It's difficult not to be impressed with how it manages to combine a tiny footprint and modest computer requirements with a sound that punches well above its weight. No, you probably wouldn't use Discover to score a mainstream TV or film project, but students might have enough here to do their mate's film school short. In addition, for working composers, sketching an idea via Discover on a compact laptop system, before Mode Switching the project over to Core or Pro for completion is an attractive idea. Indeed, when in the early

stages of writing, there is something very liberating about working with a limited palette of sounds and not having to concern yourself with stressing your host system; Discover is ideal for this.

Spitfire Audio have knocked it out of the park with BBC SO Discover; it's free, it's compact and sounds remarkable given its incredibly small size. If you look, there are other low-cost and free orchestral sounds out there but Discover sets an incredibly high bar. Absolutely a no-brainer and ideal for anyone wanting to explore the world of composing for a full orchestra. ▰▰▰

summary

Spitfire Audio have created a brilliant entry-level orchestral library for any budding composer as well as a genuinely useful compact scoring tool.

$ $49/free.
W   www.spitfireaudio.com

CUBASE 11

CREATE.
PRODUCE.
MIX.
REPEAT.

Cubase guides you on your music production journey

Do you want to start creating your own music, bring your production up to a professional level, or streamline your workflow for short deadlines? Whatever you need, Cubase helps you to reach your full creative potential. From Hollywood blockbuster composers and Billboard Hot 100 producers right through to keen beginners, the world of music production trusts the comprehensive feature set, straightforward tools and unrivaled sound of our acclaimed music production software.

The reference standard for music production software

Produce tracks from start to finish

Suitable for all genres, levels and budgets

Fast, flexible and intuitive workflows

steinberg.net/cubase

steinberg
Creativity First

Eventide Blackhole Pedal

Stompbox Effect

Eventide have put their out-there reverb effect into a compact pedal.

PAUL WHITE

One of Eventide's most popular algorithms, Blackhole, has been included in several of the company's products now, including their H9 pedal, the H9000 rackmount effects unit and their Space Reverb pedal, as well as being available as a software plug-in. It's an effect that takes reverb beyond the more familiar halls and plates to create something much larger — though it can be dialled right back to a small, resonant metal tank if you need it.

In its latest incarnation, the Blackhole algorithm's host is a sensibly compact stompbox controlled by six knobs and three illuminated buttons. Five of the knobs have a secondary 'shift' function, so these few controls can access all the functionality of the original algorithm. Five presets can be called up directly from the pedal, though you can also control it via MIDI (by plugging a MIDI to TRS cable into the EXP port), which expands this to 127 presets. Those 127 presets are also accessible using the free the Eventide Device Manager software when a computer is connected using the micro USB port. Power comes from the included 9V adaptor, or a standard pedalboard supply.

The pedal has a TRS input jack, which is switchable for mono or stereo sources, and two TS jacks for stereo or parallel-mono outs. It is switchable guitar or line-level operation, so you can use it as a DAW insert or send effect if you have sufficient I/O, without needing a reamp box, and it could hold a lot of appeal for modular synth fans too. The EXP jack can be configured to accept Expression, Expression plus Aux, Triple Aux, a MIDI Box, or a MIDI TRS Cable.

A single Aux switch in the EXP plus Aux mode enables you to control tap tempo, while in Triple Aux mode you can bank up and down and load presets. Used with an expression pedal, the high and low range of each control can be set independently, so that the pedal works similarly to the ribbon controller in the H9-based software, morphing between two sets of control settings.

One of two footswitches works as a bypass and the other as a Freeze effect, in which the reverb tail is held indefinitely. The Freeze footswitch can also be used to access presets. There are actually two types of Freeze here: Infinite continuously layers new sound on top of the initial reverb; and Freeze holds the reverb tail without adding to it, allowing you to play over it. An LED-button above the Active switch flips that switch from latching to momentary action, while the one above the Freeze switch is used when saving or incrementing presets.

The five preset LEDs also display statuses in setup mode, for example to show the MIDI channel. As you'd expect, Setup and preset management is handled most easily using the Device Manager app, but it can still be achieved using the hardware alone.

In the default ('unshifted') mode, the knobs control Mix, Gravity, Feedback, Size, Lo EQ and Hi EQ. Feedback sends the reverb output back into the pre-delay input, and the Gravity control can go positive or negative, to provide various lengths of normal reverb in one direction and reverse reverb in the other. The five secondary ('shifted') functions, accessed using an LED-button, gives access to (pre-) Delay up to 2000ms, Q, Depth, Rate and Out Lvl. The modulation is applied within the algorithm, not simply before or after it, so it doesn't sound too obvious.

With all the controls set centrally, the result is a large but more or less conventional-sounding reverb, but it doesn't take long to find settings that produce near infinite washes that still allow the dry sound to stay intact. Other settings can sound reversed or produce repeating delays washed over by reverb; perfect for ambient and cinematic composers. There are no shimmer settings and no choices of acoustic environment, so the Blackhole pedal isn't an 'all bases covered' reverb but, rather, a practical means of putting one of Eventide's best-loved reverb algorithms into an accessible stompbox format. So why yet another version of the Blackhole? Well, it's less costly than the Space and H9 pedals, more compact than the Space, and boasts more user controls than the H9. All of which makes it an attractive proposition if you want a device dedicated to this effect alone. ▰▰▰

▰ The single TRS input jack can be configured for use with mono or stereo sources.

summary

Eventide's most affordable and compact hardware version yet of their popular Blackhole reverb.

$ $279.
W www.eventideaudio.com

Focusrite Red 8Line

Thunderbolt Audio Interface

The spearhead of Focusrite Pro's Red range enjoys some light refreshment.

SAM INGLIS

Back in 2016, Focusrite's Pro division entered the market for high-end audio interfaces with the Red 4Pre. Not just a good-sounding Thunderbolt interface, the 4Pre was also a gateway to the wonderful world of audio over IP, providing both native systems and Pro Tools HDX rigs with the means to access up to 32 I/O streams from a Dante network. Focusrite swiftly followed up with 8Pre and 16Line variants, which retained the Dante connectivity whilst offering different complements of local I/O.

Four and a half years on, the original 4Pre has hung up its spurs, to be replaced by a new junior member of the range. This new interface is so similar to its predecessor that it could have been christened the 4Pre MkII, were it not for one thing: it has lost two of the four mic preamps that gave the Red 4Pre its name. So, making a virtue out of necessity,

Focusrite have called it the Red 8Line, since it does at least muster the same quotient of line-level inputs and outputs.

In most respects, the new 8Line is identical to the older 4Pre. The front panel differs only cosmetically, and the rear panel still sports two DB25 connectors for the titular line-level I/O,

> **"The similarity of the new Red 8Line to the original Red 4Pre really reflects how right Focusrite got things the first time around."**

two pairs of optical sockets that can carry up to 16 channels of ADAT-format digital I/O, and a coaxial S/PDIF input and output, plus mini-Digilink connectors for HDX, a pair of Ethernet sockets for Dante action and BNCs for word clock and Loop Sync. The 8Line's two Thunderbolt sockets use the newer Type-C connector, and Focusrite swim against the tide by actually providing a cable, which is very welcome. The 8Line's audio specifications are also unchanged compared with its predecessor, and by the time you

read this, it should be compatible with Windows as well as Mac OS.

Changes

Operationally, the 8Line is very similar to the 4Pre, so I'll refer readers who want a full description to my review of that unit in *SOS* May 2016. However, there are a few 'under the hood' improvements that are worth mentioning. The line-level inputs and outputs can now be aligned individually for +18 or +24 dBu operation, which will especially benefit those who want to use them to integrate outboard gear as DAW inserts. And the 8Line addresses a potential shortcoming of the 4Pre by allowing the local inputs to be delayed by up to 253 samples in order to time-align them with audio arriving over Dante.

One thing that has changed since the 4Pre was introduced is that these settings are no longer made using Focusrite Control. That utility is now associated only with the company's project-studio

interfaces, and local settings for Red interfaces are handled within the RedNet Control program that is used to configure Focusrite's other networked audio products. The Red 8Line control tab within this utility offers much the same functionality, and lets you configure up to eight low-latency cue mixes. Each can draw on up to 30 sources, meaning DAW playback channels and analogue, digital and network inputs. There is, however, no change to the rather basic monitor control features, so if you want speaker switching, talkback, mono and so on, you'll have to buy an additional monitor controller.

A limitation of the 4Pre at time of launch was that although it presented a total of 56 inputs, these couldn't be reordered, so if you were using a DAW that only supported 32 inputs (ie. the native version of Pro Tools), much of the Dante I/O was inaccessible. That has long since been rectified, and RedNet Control permits you to freely assign physical
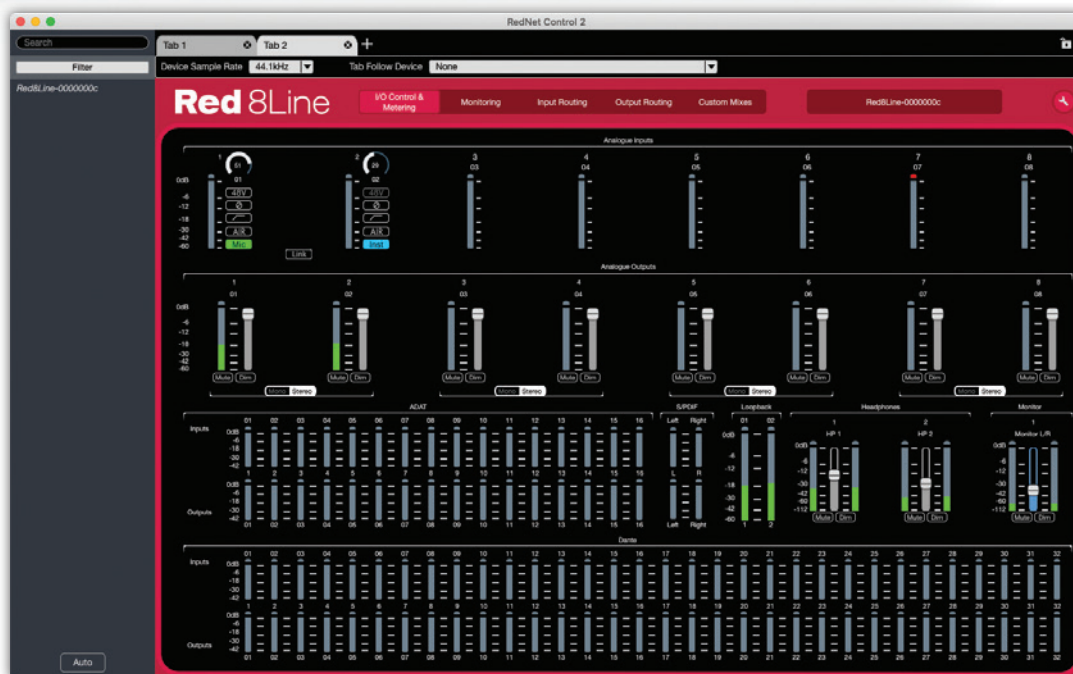
inputs and outputs to DAW inputs and outputs in any paired combination.

Conclusion

It's an example of the way in which the already impressive Red interface concept has matured since 2016, and the similarity of the new Red 8Line to the original Red 4Pre really reflects how right Focusrite got things the first time around. It's a shame to lose two of the rather nice Red Evolution mic preamps, but as I said four years ago, this interface is primarily a hub for expansion, and in most real-world installations, it'll be paired with additional

converters and preamps, or with a console. Two mic preamps is plenty for those occasional control-room overdubs. Red Evolution, in fact, pretty much sums up the Red 8Line, and with networked audio becoming ever more important, I'm sure it will continue to evolve. ▰▰▰

summary

Focusrite have brought the Red 4Pre up to date with Thunderbolt 3 ports and +24dBu alignment.

$ $2599.99
W www.focusrite.com

Novation AFX Station

Analogue Synthesizer

Novation have injected the Bass Station II with a healthy dose of Aphex acid.

RORY DOW

Anything sporting the name of IDM superstar Richard D James, aka Aphex Twin, aka AFX, is likely to get a lot of attention in synth circles. Novation got quite the scoop back in 2019 when they updated the original Bass Station II synth with 'AFX Mode' as a free software update for all users. They also added a slew of other new features since the Bass Station II's original release back in 2013.

This new limited edition AFX Station is in reality a Bass Station II with the same AFX software update plus a new look, but as we haven't covered the Bass Station II since Paul Nagle's original review back in September 2013, we thought we'd see if the AFX Station, or indeed the original

Bass Station II with all its new features, still competes in 2020.

Non-identical Twins

If you're unfamiliar with the Bass Station II, I will refer you back to Paul's original review for the gritty details (go to www.soundonsound.com/reviews/novation-bass-station-2). In a nutshell, it's a two-oscillator analogue monosynth. The audio signal path is all analogue, but the modulators are digital. There are two filters, two LFOs, two envelopes, an arpeggiator, a sequencer and 127 preset slots. Round the back you find: MIDI In/Out on 5-pin DIN, a sustain pedal input, an external audio input, a mono line output and a headphone output. Lastly, there's a USB type-B socket for USB MIDI and software editor support.

All of this remains exactly the same on the AFX Station, as does the layout of the knobs and switches. The main differences between the Bass Station II and the AFX Station are purely cosmetic, although it does come with a new preset bank from some fairly high-profile sound designers, although sadly none from Mr James himself.

The original blue and black colour scheme is replaced by purple and black, some font changes and new knob-caps. Before you run off to the classifieds to find a pre-loved Bass Station II, however, bear in mind one important thing: added to the AFX Station front panel are labels for 10 shortcuts, all new features added since the Bass Station II's first release. For example, entering the synth into 'AFX Mode' requires pressing the Function

RØDE
MICROPHONES

"The TF-5 is designed to be no-compromise… Very clean, very musical."

Tony Faulkner

T. Faulkner

RØDE TF-5
PREMIUM MATCHED PAIR
CONDENSER CARDIOID

TF-5

NOMINEE
SOS AWARDS 2021

PREMIUM MATCHED PAIR CONDENSER CARDIOID

The RØDE TF-5 is a premium small diaphragm cardioid condenser microphone designed in collaboration with legendary classical recording engineer and producer Tony Faulkner. Harnessing new capsule technology precision engineered at RØDE's state-of-the-art facility, it delivers a natural, finely detailed audio image suitable for the most demanding recording applications.

The Choice of Today's Creative Generation.™

SOS | EDITORS' PICK
GEAR OF THE YEAR 2020

rode.com

» key and the second F# key twice. These features are clearly marked on the AFX Station, but Bass Station II owners will have to memorise them.

Updates Updates...

Since its release, the Bass Station II, and by extension the AFX Station, have received a number of excellent new features. 'Oscillator error' introduces variable randomness into the oscillator pitch, to simulate the instabilities of old analogue VCOs. Paraphonic mode allows for playing two pitches simultaneously by tracking oscillators 1 and 2 independently. Envelope looping has also been added, with either or both envelopes capable of looping between 1 and 16 times. 'Fixsus' turns the envelopes into ASR envelopes and the Decay controls how long the sustain is. These envelopes always complete their full run when triggered, so are great for making drum sounds (ideal for AFX mode).

The original Bass Station II had fixed filter tracking but that was improved too, with eight selectable steps between 0-100 percent. Glide Divergence is another unique feature which alters the amount of glide time offset for each oscillator. This sounds great when used with ring mod or hard-sync patches. Microtuning has also been added, with eight preset tuning tables and the ability to create your own. And, as if there weren't enough for the aspiring mad sound professor, you can even morph between tuning tables whilst playing. Lastly we have independent sub-oscillator tuning, which means the sub-oscillator can be tuned just like a regular oscillator, opening up those lovely triad chord-stabs normally reserved for three-oscillator synths.

So What Is AFX Mode?

In short, AFX Mode allows you to make changes to a sound for every key on the keyboard. This works by enabling an 'overlay' on any patch. Overlays are a modulation source which remembers knob and switch positions for 25 different keys (the length of the AFX Station's keyboard). Outside of these 25 keys, the patch on top of which the overlay sits will play as normal.

Because every key in the overlay can hold a different value for every knob or switch, you can go from subtle per-key variations of a patch, to an entirely

Novation Components

Components is Novation's web-based editor interface for all their hardware products. Along with the release of the AFX Station comes an update to Components which allows you to edit AFX mode directly from your computer. Components also easily manages software updates plus backing up of patches, overlays and tuning tables.
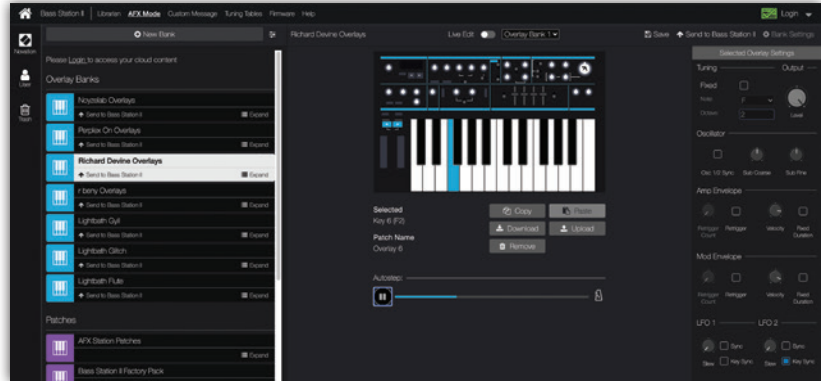
different sound on every key. There are just eight preset slots for overlays, and each of the 127 patches can reference one overlay. This means that if you change an overlay that is used in more than one patch, it will change for all patches.

What I love about this feature is that it forces you to think about modulation in a new way. Remember that this is still a monophonic synth, the overlays are simply changing lots of parameters every time you press a key. It's an ingenious idea. I found there were two main approaches to using them. You can treat each key as a separate sound and create lots of individual noises — this creates great drum kits. Otherwise, more subtle overlays can be used to create variations on a melodic sound by tweaking the sound for individual keys. I found that programming a bass or lead line, and then using an overlay to inject some variation into one or two notes was very pleasing.

Conclusion

The AFX Station is not a new product, but rather a refresh of an old one. To be clear, if you own a Bass Station II, you can enjoy all these features now for free by simply downloading the update from the Novation website. If you've been considering the purchase of a Bass Station II, the AFX Station has one very

clear advantage: the shortcuts to all the new features added since the original release are printed on the front panel right above the relevant key. If, like me, your memory is not what it used to be (too many late nights listening to Aphex Twin albums in the 1990s, I suspect), this is a bonus well worth having.

Taken as a whole, the AFX Station offers incredible value for money. Not only does it give you a great-sounding two-oscillator monosynth, but there are enough experimental features to keep even the most diehard synthesist happy. Enabling AFX mode over a paraphonic microtonal patch that morphs from one tuning table to another can make you feel like a synth god (although making it sound good is the real celestial challenge). Novation have done an amazing job keeping the Bass Station II feeling fresh with feature-packed updates, and the AFX Station is in some ways the natural end point. It is a limited edition though, so grab one while you can. ◢◢◢

"There are enough experimental features to keep even the most diehard synthesist happy."

summary

The AFX Station is a fresh coat of paint on the Bass Station II, a synth which has, thanks to continual development and free updates, and much like Aphex Twin's music, managed to stay relevant, enticing and just a little bit mad.

$ $439.99
W www.novationmusic.com

Griffon Fuzzyphone

Dynamic Microphone

Griffon's new Fuzzyphone is part mic, part fuzzbox!

CHRIS KORFF

French company Griffon Microphones make a range of small- and large-diaphragm mics, some based on vintage German and Austrian models, and some of their own design. The Fuzzyphone they sent me for review, however, is a little unusual, in that unlike the rest of their range, it's intentionally designed to be bandwidth-limited, and it incorporates a variable distortion effect.

The idea of lo-fi microphones isn't new — people have been modifying old telephones and CB radio mics for years — but in the last few years, a few companies have started offering their own off-the-peg takes on the idea, notably Placid Audio with their beautiful handmade copper mics, and Wasaphone, who make a few steampunk-inspired options, as well as a pre-modded telephone handset with XLR output.

Those mics mostly rely on outmoded capsule technology to provide their crunchy, gritty, vintage-style sounds, but the Fuzzyphone's capsule is a pretty standard cardioid moving-coil affair. Its frequency response is quoted at 140Hz-10kHz (with no qualifications, not that it matters much in a mic that isn't exactly designed for faithful capture), and its output

impedance is a nominal 250Ω. It relies on phantom power to operate, and its output is on XLR.

Aesthetically, there's something quite charmingly homespun about the Fuzzyphone, which sits just on the right side of shoddy in terms of build. Perhaps appropriately, it looks a little like a boutique overdrive pedal, and it's made rather like one too, with all the ingredients housed in an unfinished aluminium case, and a big red knob protruding from the rear, below the Fuzzyphone label and above the XLR output. The front is where the sound goes in, and you can see the moving-coil capsule through a red grille. In fact, you can see it extremely well, because it's pushed right up against said grille — which makes it easy to overload the capsule, as I discovered when plugging it in...

Hot Fuzz

Perhaps because I expected it to have a low output, and maybe also partly because of its shape, I first tested it by speaking straight into it like a walkie-talkie. Unsurprisingly, given how close I was to the mic's diaphragm, it immediately overloaded even with the distortion knob turned all the way down. With a little more distance, though, I was able to get a surprisingly clean sound out of it — not quite SM7B or Heil PR40 levels of broadcast-friendly smoothness, but not a million miles away from an SM57 in terms of clarity and HF extension.

Dialling up the distortion effect is where it gets fun. One thing to watch out for is that, on the review model at least, the knob worked backwards, with fully clockwise being clean and fully anticlockwise dishing out the most distortion. This is quite intuitive if you're turning the knob while singing into the mic, but not so much when facing the back.

On vocals, the effect ranges from clean, via mildly crackly and lo-fi (a little like an old AM radio), all the way up to Deftones-esque screaming, and producing quite guitar-like tones at max. It's also very level dependent, and therefore varies quite a lot with distance. On acoustic guitar and with the mic a foot or so away, the effect tops out at something like a mild overdrive — which actually sounded very pleasing and bluesy. On a guitar amp, you can coax wilder tones out of it by either moving the mic closer to the grille, or just turning the amp up. Sadly, I didn't have a drum kit handy to try it out on, but I can well imagine it being useful in the role of a compressed 'grot mic', and likewise as a secondary mic on a bass cabinet.

Adding distortion to sounds other than guitar is one of the worst-kept secrets in pro audio, and there are a million and one plug-ins around that are designed for the purpose. But the experience of recording with it 'baked in' is a different one, and can yield very different performances compared with adding distortion after the fact, especially on vocals. The Griffon Fuzzyphone is cheap and its build quality is pretty agricultural, but its funky looks and ability to get a unique sound out of pretty much anything could add just the bit of spark that a flagging session needs. In short, it's a fun mic to have around! ▰▰▰

summary

The Fuzzyphone has lo-fi on tap, but can also be coaxed into producing some pretty wild distortion effects. It's affordable, and every bit as fun as it looks.

$ €120 excluding shipping.
W griffon-microphones.jimdofree.com

JOHN WALDEN

Toontrack
Classic Rock EBX

Expansion For EZbass

Toontrack's debut EZbass Expansion hits all the right notes.

Having impressed on launch, it's no surprise to anyone that Toontrack are now rolling out expansion packs for EZbass. The first of these — Classic Rock EBX — is a companion to the Legacy of Rock SDX and Classic Rock EZX for Superior Drummer and EZdrummer respectively and is obviously intended to deliver that classic '60s and '70s rock sound and performance style.

Classy Classic

The instrument sampled here is a 1963 Fender Precision Bass and, while the core EZbass library includes an excellent Vintage Bass instrument (based upon a Fender Jazz Bass), sonically, the Classic Rock EBX does sit in a somewhat different space. It's warm and present and, alongside some supremely solid DI sounds, the presets favour a more rock-orientated set of amp tones with some very musical overdrive that responds beautifully to playing dynamics.

There is also a somewhat different approach to the presets reflecting the single pickup configuration so, as well as a set of finger-played presets, you also get separate sets for the pick based upon striking positions either close to the neck or close to the bridge. And while this is '60s/'70s-inspired classic rock in terms of tones, you do get one welcome nod to the more modern bass-playing world; in its virtual form, this 1963 Precision comes with five strings.

As with the EXbass core libraries, you also get an impressive array of performance articulations including up and down strokes, ghost notes,

percussive hits, tapping, up and down slides and harmonics. All of these can be easily added within the EZbass Grid Editor for extra performance realism.

You Gotta Groove

As with expansions for Toontrack's drum products, the MIDI grooves might only account for a few MB of the overall content, but don't underestimate their value. For the Classic Rock EBX, these come in a good range of tempos with a variety of straight, swung and shuffle options. Most are in 4/4 but there are also some 3/4 grooves present. They also span playing intensities, from sparse to busy, and complexity, from simple root-note patterns through to more riff-like or chord arpeggio-based grooves. Many of the grooves include chord progressions but, of course, they will all follow whatever chord sequence you specify within the Song Track.

The MIDI grooves are very inspiring to work with and, while they perfectly fit a broad palette of classic rock styles from the '60s and '70s, they could easily be dialled in to more contemporary musical contexts whose roots might lie in that classic era. Many of the grooves have suitable articulation options and chord transitions built into them; simply drag and drop a few to build a song structure, tie them in with

some appropriate Superior Drummer or EZdrummer content, and you have an instant(ish) dollop of high-class rhythm section ready to inspire your song writing. It's a heck of a workflow.

Please Sir, Can I Have More?

As with SD3, EZbass ships with plenty of sound and groove content to keep you inspired for many a project. However, the Classic Rock EBX is a very tempting addition for any EZbass users who have musical roots in this classic rock era (that's me then). The Classic Rock EBX is a great debut and, personally, I can't wait to see what Toontrack might have lined up next for the EZbass line. ///

summary

For non-bass players, EZbass is a brilliant option and, as the first expansion, the Classic Rock EBX is well worth adding.

$ $79
W www.toontrack.com

OWC Docks & Storage
Computer Peripherals

OWC make high-quality computer accessories with musicians in mind.

SAM INGLIS

Other World Computing are a long-established supplier of storage and expansion solutions for computers. One of the things that sets them apart is that many OWC products are designed to meet the needs of professionals working in audio and video. This, in turn, means the company have much more of a focus on the Mac platform than most manufacturers in their field. On the software side, they are responsible for the MacDrive utility, which allows disks formatted using Apple's APFS and HFS+ systems to be mounted and used on Windows machines, and SoftRAID, a cross-platform package which amalgamates separate drives into a RAID 0, 1, 4, 5 or 1+0 array at the software level, either for faster data streaming, redundancy or both.

The hardware division of the company, meanwhile, offers a huge variety of storage options, including internal and external SSDs, network-attached storage and hardware disk arrays. Another strong focus is expansion, with half a dozen Thunderbolt and USB 3 docks to choose from, alongside the Mercury range of PCIe expansion chassis and the Akitio Node Titan external GPU system. Other hardware in OWC's line-up includes a comprehensive selection of cables and adaptors as well as niche products such as Mac-compatible RAM and the clever Data Doubler, which replaces the optical drive in older Mac laptops with a second SSD for additional storage.

Sitting On The Dock...

It would be impractical to review everything in OWC's line-up, so as a representative sample, I chose to look at three of their products: the Thunderbolt 3 Dock, the Envoy Pro EX external SSD, and the Thunderbay 4 RAID system.

The first of these is very much targeted at Mac users who have been short-changed on expansion options by Apple. For a couple of years now, Mac laptops have sported only a limited number of sockets, and Apple have phased out USB Type-A connectors, preferring to provide Type-C sockets that can host either Thunderbolt 3 or USB devices. As they've also dumped the old and rather wonderful Magsafe power connector, this leaves some laptop users with only a single Thunderbolt socket available for connecting all their interfaces, drives, dongles, external displays and what have you.

The Thunderbolt 3 Dock goes a very long way towards righting this wrong. It's a chunky piece of kit about half the size of a Mac Mini, and requires an equally hefty line-lump external power supply. Audio interface manufacturers would do well to note that OWC provide Thunderbolt 3 cables with all products that need them, though the one included here is only just over a foot in length. This connects to one of the Dock's two Type-C connectors; the other can be used to daisy-chain further Thunderbolt peripherals or displays, and there's also a mini-Display Port socket (which, alas, does not act as a Thunderbolt 2 adaptor). One of the Thunderbolt 3 sockets is intended to supply power to an attached Mac laptop, meaning you'll only need the Thunderbolt Dock's PSU connected and not the laptop's.

This is all well and good, but the real value for many Mac users will be found in the extensive USB socketry. On one side of the unit you'll find four USB

OWC Docks & Storage

PROS
- A comprehensive range of storage and expansion products, many designed specifically with video and audio applications in mind.
- Mac users are fully catered for.
- Thunderbolt cables supplied.

CONS
- The short maximum length of Thunderbolt 3 cables means that if the Thunderbay's fan noise bothers you, there are limited options for doing anything about it.

SUMMARY
OWC's product range has been developed with the needs of Mac users and audio professionals at the top of the agenda, and it shows.

amphion®

beautifully honest

"**Amphion Two18 tells me truth**"

Rob Kinelski
Billie Eilish, ALMA, Big Sean

"**I can't work without them!**"

Jacquire King
Kings of Leon, Tom Waits, James Bay

"**They always give me the information I need**"

Dave Pensado
Michael Jackson, Mariah Carey, Christina Aguilera

"**Even at a low level you can hear so much**"

Lu Diaz
DJ Khaled, Beyoncé, Jay Z

"**I mix twice as fast, I hear things I never heard before**"

Fabian Marasciullo
Lil Wayne, Rick Ross, T-Pain

amphion.fi MADE IN FINLAND

Type-A sockets, all supporting the USB 3.1 protocol and thus backward-compatible to all previous versions. A fifth Type-A socket appears on the reverse side and is specifically badged as a 'high-powered' port, which should make it perfect for bus-powered interfaces. There's also a USB-only Type-C port, SD and microSD card slots, and an RJ45 Ethernet port.

In short, then, connecting the Thunderbolt 3 Dock to something like a MacBook Air transforms a machine that is dubiously usable for serious music-making into something that can rival any Windows laptop for connectivity. The chief cost in doing so is that you become dependent on a mains power supply, but then serious music-making rarely takes place without one. As long as you can find space in your laptop bag for it, the Dock is a comprehensive and very well thought-out solution to a whole bunch of eternal Mac problems.

▬ Thunderbay 4 is not a low point in a series of hit '80s action movies, but a RAID system.

...Of The Bay

OWC's Thunderbay 4 RAID system is available either as an empty housing, ready for your own drives to be added, or in a number of pre-stuffed configurations. The chassis itself is a pretty substantial piece of kit, weighing in at almost 4kg without drives. As the name suggests, it can house up to four 2.5- or 3.5-inch SATA 6Gb/s drives, and if you order it with disks pre-fitted, everything should be pretty much plug and play. Once again, the Thunderbay is supplied with a Thunderbolt cable, slightly more generous in length than that included with the Thunderbolt Dock.

The drives themselves are held in place by a thumbwheel of the sort that was sometimes used to fix mixer modules into large-format consoles, with a lockable door at the front. On the rear panel you'll find an IEC power inlet, two Thunderbolt 3 connections (a Thunderbolt 2 variant is also available), a security port and an HDMI connector. The Thunderbay can thus be used to daisy-chain further Thunderbolt devices as well as a display.

Also visible on the rear panel is the grille for a cooling fan, which OWC call "whisper-quiet". It's good that they are aware of the problem of fan noise in creative contexts, but in the review system, I have to say this description seemed a little optimistic. It would not be realistic to expect silence from a box

containing four hard drives, but the fan is clearly audible even when no drive activity is taking place, and seems to run at a constant speed regardless of the internal temperature. If you wanted to banish the Thunderbay to a machine room without also exiling your computer, the only obvious way to do so would be using a Thunderbolt 2 optical cable and adaptor.

Once you've installed OWC's SoftRAID XT, a version of which is supplied with the Thunderbay, the drives in the Thunderbay show up as a single volume within your operating system, and can be treated as a conventional drive. The review system was preconfigured to use RAID 5 across four 2TB Toshiba drives. In effect, the use of parity data in RAID 5 means that the price of redundancy is the capacity of one drive, so the total capacity as seen by Mac OS is 6TB. That sacrifice buys you the security of knowing that your data will survive the loss of one of the four drives in the array.

Array Of Sunshine

Some would argue that the main benefits of RAID arrays have been matched by other technologies in recent years. Modern solid-state drives are faster and more reliable than spinning-plate hard disks, whilst cloud-based backup strategies offer a level of robustness that is hard to match with physical drives. Nevertheless, storage is a numbers game,

and some of the most important numbers still favour hard drives and RAID arrays. Per terabyte, SSDs currently cost a little more or less than four times as much as 3.5-inch internal hard drives, depending on whether you get SATA or NVMe variants; and whereas individual SSDs of greater than 2TB capacity are still relatively rare, you can buy single hard drives offering 16GB or more of storage. So, although the review configuration could be matched using SSDs, 8TB is a pretty conservative spec for the Thunderbay, and the largest supported configuration offers a colossal 64TB of storage. You'd need a fast Internet connection and a lot of cloud capacity to back that up online!

According to the specifications, the Thunderbay 4 can support data transfer speeds of up to 1.5GB per second. What's actually achieved in practice will depend on the speeds of the drives that are fitted, other traffic on the Thunderbolt/ PCI bus, and the type of data that is being transferred. To get a real-world idea of the performance of the review configuration, I ran Blackmagic's Disk Speed Test, which is intended to estimate performance as it pertains to the streaming of large media files. This reported write speeds of around 160MB/s and read performance of around 580MB/s: marginal for some forms of uncompressed high-quality video, but easily enough for any multitrack audio project, and much better »

The Envoy Pro EX is a portable solid-state drive.

» than you'd get from a typical portable hard drive, especially on the read side. Orchestral composers would also have to be streaming an enormous number of samples before drive bandwidth became an issue. If it does, you could always choose a Thunderbay configuration that uses SSDs. As only the SATA standard is supported, this still won't quite give you the same performance as, say, the internal NVMe SSD in my newish Mac Mini, which measured a blistering 2.7GB/s for both read and write, but I can't imagine many real-world situations in which it would prove a bottleneck.

Bus Travel

From the user's point of view, the Envoy Pro EX is the simplest of the three review products. It is a portable, bus-powered device that connects to the host computer using a captive Thunderbolt 3 cable (USB variants are also available), and contains a single solid-state drive. The case itself is made of rigid aluminium and is protected by a chunky rubberised sleeve. Between them, these are said to provide drop protection that meets the US military MIL-STD-810G standard.

The Envoy Pro EX is available in various different capacities. Importantly, the internal drive interfaces using the newer NVMe standard rather than SATA. This, in conjunction with the high bandwidth of Thunderbolt 3, makes it much faster than most conventional portable drives: the specs quote maximum read speeds of up to 2.5GB/s and write speeds of 800MB/s. Oddly, Blackmagic's Disk Speed Test actually reported higher write speeds even than this, and suggested that streaming almost any form of raw video data to or from the Envoy Pro EX should be feasible. It's not hard to see how a portable drive that's silent, rugged and extremely fast can be of use to audio producers too, especially those with large sample collections or project libraries that regularly need to be moved from machine to machine. The Envoy Pro EX is certainly much, much more than a convenient backup option!

Other Worldly

This review represents a representative sampling from a very comprehensive product range, and has left me very impressed. From the packaging to the construction to the simple installation process for SoftRAID XT, everything about these products exudes quality, and they're backed by a three-year guarantee and 24-hour support. Some of their offerings are niche products that will be invaluable to Mac users; others, like the Thunderbay and Envoy Pro EX, are more universal in application, but benefit from a genuine consideration of the needs of audio and video professionals. Choosing the right computer peripherals will never be as much fun as buying a new synth or plug-in, but it could turn out to be a lot more important in the long run. ▰▰▰

An Introduction To RAID

RAID stands for Redundant Array of Inexpensive Disks (or Independent Disks, depending on who you ask), and the basic idea is to distribute data across multiple physical drives in a way that is invisible to the user. Depending on how it's configured, this can have two main benefits. One is speed: by writing alternate sections of a file simultaneously to two drives, a RAID array can halve the time it takes to write the entire file. Double the number of drives again and you have, at least in theory, quadrupled the data rate of the individual disks, assuming they're all the same to start with.

The second benefit is redundancy. If, rather than writing alternate sections of a file to separate drives, we write the entire file simultaneously to both, it consequently exists in two places; so, if one of those drives fails, the data is not lost. This is not in itself a robust backup strategy, since it is still vulnerable to fire, theft and other calamities, but it covers you against one of the main causes of data loss.

There are a number of different RAID standards, which balance out these two benefits in different proportions. RAID 0 maximises read/write speeds by 'striping' data across all the disks in the array, but offers no redundancy at all, whereas RAID 1 'mirrors' all data to all the disks in the array, thus offering maximum redundancy with no performance benefit. For many applications, a good compromise is found in RAID 5. This requires at least three drives to implement, but, in effect, 'stripes' your data across as many disks as possible whilst retaining some redundancy through the use of a system called parity.

The job of distributing data across multiple disks, and presenting them to the OS as a single entity, is handled by something called a RAID controller, which can be implemented in hardware or software. An array that uses a hardware RAID controller should, in theory, be indistinguishable from a conventional drive as far as the computer is concerned; you can plug it into any machine and it'll immediately be recognised as a single volume. A software RAID array requires the relevant software to be installed on every machine that wants access to the drive.

Both Mac OS and Windows actually include built-in software RAID controllers, but the Mac OS implementation only supports the basic RAID 0 and 1 standards. For this and for a number of other reasons, more sophisticated third-party software RAID packages are widely used, and OWC's SoftRAID is a good example.

Win!
Austrian Audio OC818 Dual Set Plus

Worth $2199

Austrian Audio are a relatively young company, but the people behind it have some serious form when it comes to making microphones. The brand was founded by a group of AKG engineers who, in their spare time, had set themselves the challenge of creating a microphone capsule that had all the magic of the venerable brass-ring CK12 — as found in AKG's classic C12, the original C414, and Telefunken's ELA M 251 — but that could be manufactured consistently and cost-effectively (a perennial problem with the original design).

The fruit of their labours was the CKR12. Handmade in Vienna, it's this large-diaphragm, Braunmühl-Weber capsule that lies at the heart of their flagship OC818 multi-pattern microphone.

When *SOS* Editor In Chief Sam Inglis reviewed the OC818 last year, he was hugely impressed by its performance, describing it as "beautifully clean, with a supremely balanced sound". But it wasn't just the sound that set it apart: the OC818 has the unusual feature of a secondary mini-XLR output, tapped from its rear-facing diaphragm. By recording both diaphragm outputs separately, you can change the resulting polar pattern after the fact in your DAW, using AA's PolarDesigner plug-in.

For this month's competition, Austrian Audio are giving one lucky reader their newly announced Dual Set Plus, a stereo kit that comprises two of their premium OC818 mics plus a host of accessories, all shipped in a rugged aluminium case. A pair of windshields is provided, as are two standard mic clips plus two of AA's own shockmounts, designed specifically for their OC-series mics. You also get a pair of mini-XLR adaptor cables, so you can easily take full advantage of that secondary output, as well as an adjustable stereo mounting bar.

To be in with a chance of winning this fantastic prize, all you have to do is follow the URL shown, and answer the questions there, by Friday 5th February. Good luck! ◢◢◢

Prizes kindly donated by Austrian Audio
W austrian.audio

To enter, please visit:
https://sosm.ag/aacompjan20

Legendary Ocean Way Audio sound, now affordable for your home studio or project room. HR5 Reference Monitor, now available with new S10A Subwoofer.
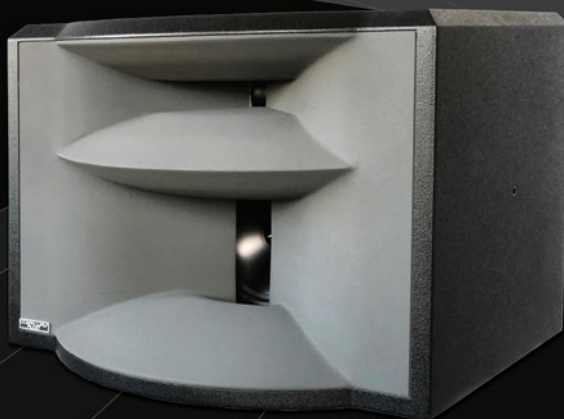
HR5

- The most affordable Ocean Way Audio loudspeaker ever made, excellent for home studios and production rooms

- A perfect 100 degrees of horizontal dispersion for consistently great sound from anywhere in the room

- ±1db of matching performance throughout their entire extended frequency range

- Clear and defined bass down to 45hz and below… remarkable for a small speaker

S10A

- A prefect companion for the HR5

- 20Hz-120Hz frequency range with 110dB maximum SPL

- Stereo XLR inputs; flexible controls for gain, LF adjust, crossover, delay and more

oceanwayaudio.com

OCEAN WAY® AUDIO

Zivix Jamstik Studio

MIDI Guitar

Some of the performance issues associated with
guitar-to-MIDI systems may not be insurmountable after all!

DAVE LOCKWOOD

Minneapolis-based Zivix build innovative electronic instruments and related software, including the Jamstik range of 'guitar-like' designs, created with an emphasis on portability and learning. The latest model in the Jamstik range, though, the Jamstik Studio, is very much a real guitar, utilising a Chinese-made, compact body, headless design from ALP Guitars, and incorporating within it one of the best pitch-to-MIDI-based systems yet made. Yes, you read it right: it's pitch-to-MIDI and it's really rather good!

Pitch-to-MIDI conversion — reading the vibration of strings on a real guitar and endeavouring to convert the pitch and amplitude information to MIDI data — has been at the heart of most MIDI guitar systems that have made it to market in significant numbers. Roland especially deserve an honourable mention simply for sticking at it for decades, and there have been incremental improvements over the years, notably from Axon and latterly the Fishman TriplePlay system.

The inherent limitations of pitch-to-MIDI, namely pitch-variable latency and mis-tracking, have spawned a number of new approaches to MIDI conversion, from the wired frets of the Zeta Mirror 6, the Sonar detection of the Yamaha G10, through to instruments like the Synthaxe, the Starr Ztar and You Rock Guitar. Some of the latter group actually work quite well as MIDI controllers, mainly because they are not guitars at all, but really 'keyboards' whose notes are arranged in a guitar-like pattern, albeit with the complication of the involvement of secondary triggering with the picking hand.

Two Tribes

It seems to me that two distinct groups of people take an interest in MIDI guitars. The

techniques they normally rely on. In reality, guitar-to-MIDI systems don't actually serve these people particularly well either, and it's not surprised me to see a growing trend in recent years to bypass MIDI altogether and use digital oscillators and filters acting upon the signal taken directly from the strings; this makes for very playable synthesized sounds, with good tracking and no latency. But it doesn't get you MIDI and all the advantages it brings in terms of the range and quality of sounds available and the deep editing available in DAW software.

Remarkable

Just when we all thought pitch-to-MIDI had nothing substantially new to offer, the MIDI guitar world was shaken up by the arrival of the Jam Origin MIDI Guitar app. This remarkable piece of software utilised the DSP of its computer host to analyse the output from a normal electric guitar signal with all the strings combined into a single audio stream. It wasn't significantly faster than other pitch-to-MIDI systems, in my experience, but it was reasonably clean and accurate, although to many people, still the most remarkable thing was that it worked at all.

It did set my mind to wondering, however: if today's DSP could now extract MIDI out of the combined waveform of a polyphonic guitar signal, what might it be able to achieve with a hex pickup that gave it each string as a separate signal? Well, that's rather what I think is going on here in the Jamstik Studio. In this case, the DSP is on board the guitar and because the system is integral, rather than a retro-fit, the designers benefit from the elimination of a number of possible variables: the exact relationship between the pickup and the strings is known, as are the bridge type, the scale length and the resonance of the body. Having to allow for variation in all of these factors in retro-fit systems makes a difference to how hard the software has to work just to do the basics, and if it has an easier time doing that, it can devote more power to trying to interpret your playing in meaningful ways. The curious thing about the Jamstik Studio compared with other systems is that there is nothing to adjust in terms of playability parameters — it just works!

Setting Up

The Jamstik Studio guitar has a 'headless' configuration with a scaled-down mahogany body, somewhat like a travel guitar, but it sensibly retains a full 25.5-inch scale-length

on a 24-fret, bolt-on neck that is a familiar 42mm wide at the nut. A pair of covered humbuckers, three-way selector switch and volume control with 'pull coil-tap' allow it to function as a conventional electric guitar too. It performs that job adequately, if unspectacularly, and that's not intended as any kind of slur: highly characterful electric guitars don't make the best MIDI platforms, in my experience. A conventional hex pickup sits behind the bridge humbucker.

Strings are inserted at the nut end of the fingerboard, leading down to the bridge: it's a slightly odd affair, comprising roller saddles, normally associated with a floating trem bridge, and what looks like thumbwheels for fine-tuning. I find them a bit too stiff and close together for finger operation, and I guess you are primarily meant to operate them with the mini hex wrench that's conveniently attached magnetically to the body of the bridge, so you won't lose it easily.

Given how important accurate tuning is to the performance of MIDI guitar systems, experienced users tend to do rather a lot of it, and whilst every attempt has been made to minimise the potential for slack and slippage in the basic design, the greatest variable of all is temperature. The pitch of steel strings moves over time as they warm up. Just playing the instrument warms them up, so accurate tuning is not a set-and-forget activity. The thumbwheel tuners' gearing is also a bit on the coarse side, I find, so adjusting them with the supplied 20mm right-angle hex key can be a bit hit and miss. I used a longer key to give me better control, but if there is ever »

first comprises guitar players who simply want to be able to play keyboard sounds from the familiar interface of a guitar. The second consists of those who want to bring some of the expressiveness of their chosen instrument to the world of keyboards, and especially to synths, where you may not be trying to replicate traditional keyboard sounds and playing styles. The two interests are quite distinct, almost to the point of being in conflict.

The former camp, in which I count myself, want the kind of 'clean MIDI' that you get from a keyboard, and for which the 'language' itself was invented. Note-On, Note-Off, Velocity value and a bit of sustain pedal gets you 90 percent of the way there, with a data density that any DAW can handle and any voice module or virtual instrument can understand.

Those in the latter group, however, require something a lot more guitar-like, with proper strings to bend and apply vibrato to, and to damp with the right hand, using all the instinctive 'guitaristic'

▬ Having the pitch-to-MIDI processing in the guitar, rather than running on a separate computer, means you can control standalone synths directly using MIDI.

» a Mark II of the Jamstik Studio, I hope it has some easier-to-use tuners.

As supplied, the guitar was adequately set up, if a little on the conservative side. MIDI guitar setup is a delicate balance of playability versus clean notes for good pitch detection. A higher action will give you a cleaner note with less fret choking, but is more difficult to play, resulting in more 'release' noises as you move about the fingerboard, all of which are usually translated as spurious short MIDI notes. The truss rod already had the ideal amount of relief for the supplied .010-.046 gauge strings, but as the nut is height adjustable, I found it could be lowered a useful amount to achieve a much better playing experience without causing any triggering problems.

Any adjustments at the bridge need to take into account the fixed curvature of the hex pickup. The latter has height and tilt adjustment via a screw at each end, and would, ideally, be positioned with its six pole-pieces at an identical distance from each string. There are sensitivity settings in the Jamstik Creator software app, but I find it always helps to start from as even a response as possible in the basic setup. Bridge adjustment is not

immediately obvious — there are two hex screws in each saddle, like a Strat, but one does the height and the other locks the position after adjustment.

The setup wasn't 'wrong' as supplied, but a bit of basic 'fettling', certainly made it more playable in a clean, MIDI-friendly way. If you make significant changes to the action, you may well find you need to reset the intonation. The saddles are locked in place by a hex bolt that sits directly under the string, which has to be slackened off to access it. Though not ideal, you should only have to do it once unless you change string gauge.

Making Connections

Digital connectivity comprises a USB-C port, used for both MIDI and charging the onboard battery, with the alternative of a 3.5mm TRS 'conventional' MIDI output. There's also the option of wireless MIDI connectivity using the BLE (Bluetooth Low Energy) protocol. This will add a little more latency to an already very latency-sensitive activity, so a wired connection is always preferable, but Bluetooth BLE is certainly a convenient option for using Jamstik Studio with iOS and other mobile devices.

Operation via the USB-C connector requires a data-enabled USB-C to USB-A cable: a 'USB-C to C' charging cable will not transfer MIDI data. The onboard Lithium-ion rechargeable battery will give in excess of

eight hours unconnected playing time, but if you are connected via a powered USB port, you'll be charging as needed anyway. I must just add, having poked around under the hood, ultimate kudos to Zivix for what looks like a user-replaceable battery. No rechargeable device should be able to be sold without this possibility, in my opinion.

The USB-C and mini-TRS, along with an activation switch/indicator, appear on a small mounting plate on the lower bout. When playing the guitar seated, as I and I suspect many others tend to do when recording MIDI into a DAW, this places the USB-C connector directly sticking into my right thigh, and I'm sure sustained usage would damage the connector over time. A right-angle USB-C adapter made it both more comfortable and, I suspect, in the long term more reliable.

Notably absent from the Jamstik Studio, is the 13-pin, multichannel analogue output traditionally associated with hex pickups. It is entirely logical, given that the Jamstik Studio is processing the hex pickup signal on board to output MIDI directly, but it does mean you can't take full advantage of 'guitar-synth' products that can use the string output directly for modelling and harmonic synthesis, like the Roland VG and SY units.

If you are wondering where you set up things like the basic modes of operation (single channel versus multichannel; pitchbend on or off etc.), that's all in the »

» Jamstik Creator Mac/Windows plug-in. That's also where you can adjust individual string sensitivity, MIDI tunings/transpositions and check tuning. It's also a VST instrument that makes good use of Jamstik Studio's MPE (MIDI Polyphonic Expression) mode, which is now part of the MIDI spec.

MIDI guitar players have been using a sort of version of MPE for years, with each string on its own channel so that individual pitchbend messages to be transmitted, but the full MPE spec allows for other per-string real-time control data, too. When the Jamstik Studio is set to MPE mode, MIDI CC11 data is generated from string amplitude, tracking the envelope of the notes and then mapping that data to the synth voices in the plug-in. It makes for a very natural 'guitaristic' playing experience, as instinctive things like left-hand lift-off damping and right-hand palm muting have a useful effect on the output. Of course, the receiving instrument has to know how to do something useful with CC11 data, so you'll not find it particularly effective just firing a torrent of MPE data at any of your DAW's virtual instruments that aren't set up for it.

Standalone

Because it has its own DSP on board, outputting MIDI directly, the Jamstik Studio is capable of standalone operation connecting to any MIDI device without a computer or tablet in between. The makers Zivix call it "the first contained MIDI guitar solution available for a mass-market audience", although I have a Casio MG510 sitting here as I write that also outputs MIDI directly. I do think Jamstik Studio would be more effective in standalone mode if it had just a little more control of basic functionality on board, specifically a 'multi/single-channel' switch, and pitchbend On/Off. As it is, you'll still have to dip into the plug-in for those — personally I'd happily dump the coil-tap switching in favour of MIDI pitchbend On/Off. The Casio had all of that on board in 1988, and a tuner, too. The Jamstik's standalone tuning can be taken care of with a clip-on (most will fit and function perfectly well on the vestigial headstock), or using the guitar quarter-inch output jack connected to a normal electronic tuner.

When you are using the plug-in for tuning, you will have to remember to have pitchbend activated as the software will be reading the MIDI output for this function. It has to, of course, because there is no analogue output from the hex pickup as there is with the 13-pin protocol. You can still use the software to tune in single-channel

▬ The outputs include a USB-C port for connection to a computer, a TRS minijack for MIDI, and a quarter-inch output for the conventional guitar signal. There's also Bluetooth connectivity.

mode once you realise that only the top-E display works for all the strings. Logically, the sensitivity monitor display works in the same way — one for every string in multichannel mode, or all strings in the top-E window in single-channel mode. Less logical, to me, is the sensitivity-adjust sliders being in a different window to the sensitivity display. There's one for each string and a master slider, and at present, adjusting the master resets all the individual sliders to match. A more logical behaviour would be that, having set your individual string sensitivities, you could then adjust the overall sensitivity up or down just to match the requirements of the synth voice you were addressing, without losing the offsets. Some synth voices benefit from a bit more push, others a bit less, while the individual sensitivities are about string-to-string balance, which, once set, you wouldn't want to change.

Currently, there is no facility for changing the preset pitchbend range parameter in the control software settings, although I understand that is about to be addressed with an update and may well be in place by the time you are reading this. Of course, you can change the setting in the receiving instruments, but they may be already set how you want them to work with existing sources.

The guitar has no onboard sound creation of its own, but the free Jamstik Creator plug-in does offer up some beautiful synth sounds that really show off the potential benefits of MPE mode. When you launch the plug-in it suggests that you

engage MPE mode for "best performance". That would certainly be the case for live use, where the possibility of greater physical interaction with the sounds certainly gives an increased sense of control. I wonder, however, if the biggest potential user group for Jamstik Studio may actually be people who just want better, cleaner MIDI for playing keyboard parts in their DAW, and they will neither want nor need to be in MPE mode most of the time. For those people, a simpler control app without the synth would be a more useful thing, addressing mode selection, pitchbend range, tuning and sensitivity setup.

Simply The Best?

Basic audio-to-MIDI conversion in software is now a well-established process that works very effectively. 'Real-time' pitch-to-MIDI, however, is a considerably greater challenge, and deriving 'real-time' MIDI from guitar playing is another layer of complexity beyond. Leaving aside the fact that what we call 'real-time' here can never actually be real-time — at least some of the note has to be in the past before you can begin working out what it was — guitars have the complication that notes can be initiated in two different places, and in fact usually are. When you move from one fretted note to another and pick the string to sound it, you will have 'pre-fretted' the new note before your picking hand goes into action. It's something we all learn very early on as players and never have to think about again. Potentially, that pre-fretting results in a low-velocity MIDI note just before the wanted picked note. Of course, that 'unwanted' note could potentially be »

CRANBORNE
AUDIO

CAMDEN

MOJO

SIG

MOJO STYLE
CREAM

OFF

MAX

THUMP

HPF

HI Z/
LINE

CAMDEN

AWARD-WINNING SOUND
Camden preamp family

The cleanest, most transparent preamp in your arsenal,
or the most colourful and characterful. Your choice, at
the turn of a dial.

Either way, you'll be bring award-winning sound to your
microphone, instruments and mixes.

excluded by applying a tiny amount of time-windowing to see if a low-velocity note is followed immediately by a picked note of the same pitch, but now we are adding a fraction more time to the conversion process over and above just working out the pitch. And some pre-fretted notes are actually wanted hammer-on notes that would never be picked anyway... and is that a hammer-on modulation of an existing note with pitchbend or an entirely new note? As you can see, it's complicated, and I have to express my admiration for people who continue to push boundaries in this comparatively narrow field.

There may be a few things about it that I would prefer to have seen addressed in a different way, but Jamstik Studio is still one of the best guitar-to-MIDI instruments I have used. It offers fast, accurate MIDI, with less spurious MIDI 'chaff' in the output than most other pitch-to-MIDI-based systems on the market today. Oddly, it 'feels' faster than it measures, although perceived MIDI-guitar latency is substantially determined by buffer size and the attack characteristics of the actual synth voice you are addressing. My measured times from pick transient to the start of the converted MIDI note have the top-E at a rapid 5-6 ms, with low-E at around 25-30 ms. The vagueness in the numbers stems from the fact that my measured times weren't always consistent, There is perhaps some contextual interpretive 'intelligence' in play. The developers are, perhaps understandably, unwilling to reveal anything about their conversion algorithm other than to say that it is new and developed entirely in-house. Whatever they've done, it works, and means that you seem to get more of what you intended and less of what you didn't, like open strings ringing, lift-off re-triggering and the rest of the rubbish that MIDI guitarists have just become used to hunting down and deleting in their piano-roll editors.

Even for the most experienced MIDI guitarist, the activity has always been a mixture of delight and frustration. Whilst there is still room for improvement, the Jamstik Studio certainly brings me more of the former than the latter, and represents a very worthwhile advance in its field. A full-size MIDI guitar has apparently "always been a dream of the team of developers at Jamstik. The product we've wanted to launch since the beginning", and I'm delighted they stuck at it! ///

$ $799.
W www.jamstik.com

The free Jamstik Creator app is used to change hardware settings and doubles up as a playable instrument.
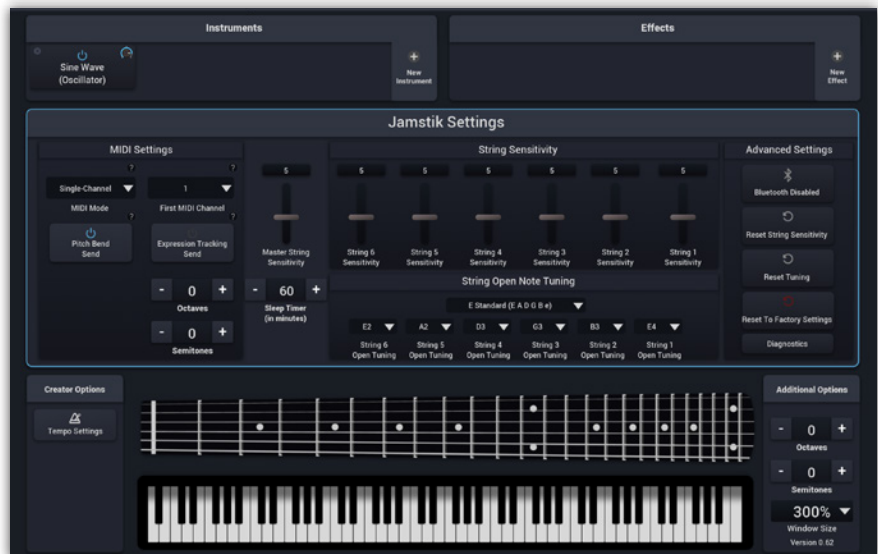
Native Instruments
Maschine+

Groove Workstation

Is the Maschine+ the standalone music-making
box you've always dreamed of?

SIMON SHERBOURNE

The arms race of beat production workstations has come full circle. In 2009 the first Maschine took cues from Akai Pro's MPCs, but by moving all the hard work to an attached computer NI opened up the scope of what was possible. Maschine had access to processing power and plug-ins, and could slot into the DAWs that had become the primary workspace for many producers.

Akai found themselves on the back foot, and the previous year's Renaissance was to be the last of the classic standalone MPCs. The next MPCs followed Maschine down the hybrid route, with a range of hardware controllers acting as front-ends to a Mac/PC-based MPC app and plug-in.

In 2017 Akai re-took the initiative and capitalised on the trend for computer-free composing and performing. The MPC Live and X saw the MPC software ported back to the hardware, using System-on-a-Chip technology to combine some of the best aspects of a computer and a standalone groovebox. Maschine+ follows suit, and is Native Instruments' first non-native instrument. So what is gained and lost in the move to mobile, and who is this for? Let's find out...

Overview

Maschine+ is a reworked version of the Maschine MkIII hardware that can detach from your computer and run standalone. It has an embedded processor running the Maschine software, with on-board storage and an SD slot for user content. Standalone operation is the same as using a regular Maschine controller, without the computer-based UI.

Rumours had suggested a standalone Maschine would be based around the MkIII, using its on-board audio and MIDI interfaces. I'd guessed it would have Maschine's internal suite of effects and its sound sources: namely sample-based kits, sample-mapped instruments, the drum synth engines, and the Bassline monosynth. I'd have been happy with that as an initial feature-set, but there are some exciting surprises. NI have ported several engines from their VST plug-ins, giving you access to NKS-format sound and effect patches. And they've enabled expandable connectivity, with USB MIDI hosting and support for external MIDI and audio interfaces.

Metal Maschine Musik

The extra brains crammed inside the Maschine+ only make it about 10 percent heavier than the MkIII controller, but it feels significantly more substantial because the front and rear panels have been replaced with gorgeous anodised aluminium. It has that smooth gunmetal finish that you find on high-end hi-fi gear.

The eight touch-sensitive encoders have had an upgrade. They are now larger, all metal and have a damped action that doesn't suffer from the variance I've noticed on my MkIII and Komplete Kontrol keyboard. They really add to the overall luxury feel. The displays are the same (there's no touch upgrade), and you get the same wide, super-sensitive pads.

Most of the rear panel is unchanged, housing all connections to the outside world including the line and headphone outs, and the mic/line ins, all of which have a tiny gain knob. However, the power button has moved to the front, and in its place is a pair of USB A ports for connecting devices when running standalone.

Firing Up The Maschine

My virgin Maschine+ booted to a welcome screen and asked for a Wi-Fi connection. A nifty QR scanning process connected the unit to my Native Instruments account and I was then able to install the core plug-ins and factory libraries, plus the five included Expansion packs. Updates were delivered over the air, and the additional Expansion Packs I bought on the NI website immediately appeared for direct download on the Maschine+. The core package comprising OS, plug-ins and sound library all live on hidden internal storage, while Expansions install to the removable SD card.

As a Maschine MkIII user who tends to stay focused on the controller I had a head start. Using the Maschine+ proved practically identical to this apart from having an additional Settings page for managing preferences, connections and installations. If you're used to doing some tasks like note editing on the computer you'll need to learn the hardware equivalents. With practice these operations can become fast and precise, and quite a satisfying way of working. The UI is

»

Native Instruments
Maschine+
$1399

PROS
- The full sampling, performing and composing experience of Maschine, without the computer.
- Incredible sonic arsenal.
- Solid build.
- Support for USB audio and MIDI interfaces.

CONS
- Sharing projects and content with desktop is clunky.
- Plug-in control via pre-mapped macros.
- No CV.
- No internal power source.
- New Clips functionality adds complexity.

SUMMARY
With desktop-class plug-ins and expandable I/O, Maschine+ pushes the boundaries of what you'd expect from a standalone workstation.

On the back panel we find stereo audio I/O, controls for headphone volume, line out volume and input gain, a pair of full-size MIDI ports, and three USB ports.

mostly smooth, and the pad response is super snappy. The audio settings screen reports a combined processing and output latency of under 6ms, and throughput monitory delay of 8.7ms. To match that on my Mac I'd need to be running around a 64-sample buffer or smaller.

Maschine has a working structure that I really like. Tracks ('Groups') are selected from the eight backlit pads on the left, and each have 16 individual polyphonic sequencing lanes and audio paths. Drum Kits populate a whole Group with 16 sounds, and can mix sampled and synth sources. The kits that come with the Maschine Library and Expansions are beautifully crafted and never fail to inspire.

The individual Sound lanes in a Group can also host key-mapped instruments or plug-ins. Maschine users will be familiar with the six multi-mode Drum Synth modules that come as standard, as well as the internal Bass Synth. Like the internal effects modules, these are native to Maschine and have no graphical interface. Modules and device chains are represented as thumbnails, and parameters are mapped in pages across the eight encoders.

Sounds

The beating heart of the Maschine and Komplete platforms is the sound browser, which collates your sound sources into a tagged patch library with audio previews and controller assignments. The desktop Maschine can run all of NI's plug-in instruments and effects, each of which has its patches in the Maschine library's NKS format. A number of third-parties like Arturia, Softube and Waves also support the NKS patch format.

Unsurprisingly, Maschine+ doesn't have general plug-in support, but it does come with a handful of NI instruments and effects over and above Maschine's internal modules. On the instrument side you get FM8, Monark, Kontakt, Massive, Prism, Reaktor and Retro Machines. Premium effects are the reverb Raum, and phaser Phasis. Reaktor, Prism and FM8 can also be used for effects, the latter

having some useful amp and cabinet effects if you plug in a guitar.

Both the Reaktor and Kontakt installs come with quite a limited subset of their factory instruments, but the key thing is that they're the engines for other packages. Retro Machines, for example, is a Kontakt-powered instrument, and Monark and Prism are Reaktor-based.

This all adds up to a huge palette of sound and synth engines compared to any similar workstation. There is a notable limitation though. Maschine+ has the same control and display access as the controller in a regular Maschine setup, so can't access things that rely on the computer screen. This includes plug-in UIs and automation data. You can only control (and automate) the parameters pre-mapped to encoders in the NKS patches. What you have then is essentially a sophisticated preset and macro synth, similar to NI's long-gone Kore or Arturia's Analog Lab. It's brilliant for tweaking and modulating existing sounds, but deeper programming or control mapping needs to be pre-configured on your computer.

User Content

Though there's a lot built in, you'll want to get your own sounds on board. This can be in the shape of pure sample content, or your own NKS-format patches. Sample-wise, the 64GB SD card should go a long way, but if you need more space it's not expensive to upgrade. The Maschine+ has a dedicated storage mode which mounts the SD card if the unit is connected to your computer. (It doesn't do this in Controller Mode but we'll come back to that.) More conveniently, you can pop out the SD card, which I found you can do even while a standalone project is open.

You can dump raw samples and loops onto the SD card, as long as you put them in a pre-defined User Content folder. The Maschine+ currently has no generic file and directory manager. However, in Maschine's Browser, your folders show up as Banks in the user One-Shots and Loops categories. I'd previously tagged most of my samples

in the desktop Maschine software, and this carried through to the standalone so I could filter by sound type.

Your own Maschine and Komplete Sounds and Projects can be moved across, again by observing correct folder placement. NI's software suite uses centralised locations for storing factory and user patches/projects. This hierarchy is maintained on the Maschine+ SD card. Moving content between the two environments is a manual drag-and-drop process. A better way to manage or synchronise your stuff is top of my wish list.

Once you figure out how the library is structured a couple of sneaky (and unsupported) possibilities present themselves: bringing in Kontakt libraries and Reaktor patches. The former is only possible with unencrypted Kontakt presets; most commercial libraries secure the sample media in an archive that the Maschine+ can't access. Kontakt libraries that do work get the benefit of disk streaming. Most hardware samplers, including the MPCs, always load samples into RAM, so are limited by the on-board memory. Kontakt (but not Maschine's native sample engine) is able to work directly from the SD card so can open large patches.

If you have a license for the full version of Reaktor on your NI account, instruments ('Ensembles') can be copied on to the SD card and show up in the Browser with their embedded patch states. This is brilliant news if you build stuff in Reaktor, but I was also able to bring in a number of the instruments from the Reaktor Factory Library, including some all-time favourites like Subharmonic and Steampipe. Reaktor Blocks content does not work. My excitement was dampened when I encountered some issues: the ensembles I smuggled in could stop responding after a while, and did not work after re-opening a project. Still, it shows the potential of this platform.

Performance

Like a computer-based DAW, Maschine+ will let you keep adding tracks, instruments

and effects until you run out of processing or memory resources. The point at which audio rendering starts to glitch out depends largely on which plug-ins you use. Projects mostly built with Maschine's traditional sample-based kits and instruments can comfortably grow large. When using the premium instrument plug-ins or a few complex effects chains, I found I could max out the CPU pretty quickly.

A typical project for me might include a drum kit, a couple of tracks with chopped samples or loops, three or four synths and a couple more tracks sequencing external gear with effects on the channels. I always add some Performance FX on the Groups as well. These kinds of projects, typically around eight Groups wide, proved generally within the Maschine+'s scope, although sometimes only just. I found high polyphony on the fancy synths like Massive or Prism, or recalling Lock snapshots (especially with Morphing enabled), were the fastest way to spike the processor and get crackles.

Spec-wise, the Maschine+ has 4GB of RAM, double that of the Akai workstations.

Power Points

The Maschine MkIII runs happily on USB power: the screens are bright enough on USB-power alone that the PSU isn't worth the trouble. The same goes for the Maschine+ in controller mode, but standalone operation requires you to plug in. The power port has an unusual locking connector that twists into place. On the face of it this is a great idea. Without a battery, it's game over should your cable get yanked out on stage. For studio use I'd rather cables detached gracefully without pulling my device off the desk. You don't have to engage the lock though, so it's up to you.

It's a shame the Maschine+ doesn't have its own battery, but you can run from external power packs which have a 15V DC supply. I tried two different battery packs designed for laptops and both worked well. I settled on a standard Sony camera battery with a 15V DC output that happily powered the Maschine+ all day. On a whim I tried the small 12V bank I use for Elektron devices and even that worked. The Maschine+ has clearly been engineered to tolerate a range of power, and in typical operation uses a lot less than its official rating.

Rather than ARM though, the CPU is taken from the Intel Atom range of embedded mobile processors. This might not have much raw grunt, but makes sense. Last year I tried to make a DIY standalone Maschine by pairing a MkIII controller with my Microsoft Surface Pro. This had plenty of power, but the fan noise and audio latency were horrible. The Maschine+ processor is modest by comparison, but it doesn't need a fan, and as its custom Linux OS is entirely dedicated to Maschine

it can run reasonably complex projects with low latency.

Clips

The established workflow in Maschine is based around Patterns, which are Group-level sequence chunks. Patterns are organised into combinations called Scenes in the Ideas workspace, where you can perform impromptu arrangements. Maschine projects also have a linear timeline where Scenes can be arranged as Sections.

The Maschine+ measures 322 x 301 x 51 mm and weighs in at 2.5kg. Many of us will have owned laptops heavier than that.

Back when we reviewed Maschine Jam, NI announced that they were working on a Song Layer that would allow free recording or placement of content into the timeline, breaking out of the grid structure. This never materialised as a Jam-powered feature, but has been resurrected as Clips in the Maschine+ software release. Clips can be recorded directly into the timeline, or you can convert Patterns to Clips. Clips and Pattens can exist on the same track, but where they overlap the Clips take precedence.

The Clips implementation affects Maschine's operation at a fairly fundamental level. This is reflected in some naming changes on the mode buttons: Arranger is now Ideas/Song; and Pattern is now Pattern/Clip. By default, projects act as they did previously: The Pattern button brings up the pattern pool for the focused track, and the Ideas button displays the Scene manager. Shift-push Ideas and you'll switch to the timeline Song view, which now has mode buttons for shifting the whole workstation between Section and Clip focus.

I found the change frustrating at times, particularly the fact that the Ideas/Song button doesn't toggle you between the modes. Once you've moved to Song mode,

the only way back is a second-layer softkey function in the timeline display view.

Although I was sceptical about the conceptual implementation of Clips, they are really useful. It's refreshing to be able to record MIDI directly into the timeline as an alternative to building from loops. It's similar to the common MPC approach of working in a single long sequence, scooting around the timeline with the main encoder. I even ended up making a completely off-grid ambient piece with no Patterns at all.

The main idea is that you'll use the Ideas and Sections concept to sketch an arrangement, then use Clips to add linear sections, fills and transitions. It doesn't particularly work for overdubbing fills

though, as Clips always mask whatever is underneath them. You can Duplicate the track, but this is wasteful of resources and splits your mix. Converting Patterns to Clips doesn't help as the whole section (every track) get converted and you get separate Clips either side of the grid boundary.

What's still missing is the ability to capture an overall performance into the Arranger. You can't lay down a Scene arrangement in real time, you can only record one track at a time, and you can't record automation or Lock snapshots across the timeline. Akai have this nailed in their latest Force update, so the challenge is set!

Audio Unit

The other thing you're inevitably going to want to do is record audio as Clips. Audio recording in Maschine is done in its dedicated Sampler mode, which captures one external or internal audio source at a time. This is fine for editing, slicing, layering and key-mapping samples for use in the Sampler instrument plug-in. For straightforward playback of loops or linear audio, Maschine uses an Audio plug-in in

its MIDI tracks. This simply loops one clip per Sound lane, and gives you the option to gate and pitch the audio via MIDI.

This is OK for dropping in loops, but is super fiddly for actually recording audio into a Song. There's no equivalent to the audio tracks in a DAW (or MPC), where you can simply record into the timeline. Clips does give you an approximation: you can arm the Sampler in Sync record mode, create a clip in the Arranger and record into it. But each new recording requires you first to switch to a new sound slot in the track and load an audio device onto it. It's confusing and frustrating.

Maschine Jam had a built-in workflow that pulled the functionality together in a usable way, letting you quickly grab audio into Pattern slots like a looper or Octatrack. This is not present on the Maschine+ and frankly Maschine now really needs a straightforward way to record and manipulate audio in the timeline.

Peripheral Vision

Like the MPCs, the Maschine+ hosts MIDI devices directly over USB, and even

supports MIDI interfaces: I connected a Retrokits RK-006 and gained 10 extra output ports! NI's keyboards work as generic MIDI controllers, and the MkII S-Series, A-Series and M32 will eventually get extended support. This will bring the same browsing, control mapping and smart play features as with the Komplete Kontrol and Maschine desktop apps.

External MIDI synths can be sequenced over USB or standard MIDI ports, and clock can be received or sent over either. MIDI CC control is somewhat problematic from a standalone point of view because you can't create the necessary macro controls directly on the hardware. You need a Sound preset to have been created in the desktop app. Native Instruments have included templates for some popular devices with extensive, labelled controls.

I was particularly interested in whether I could connect my Maschine Jam. Jam's easy Pattern, Scene and Lock access, and mixer and effect controls, save you a lot of mode switching, and are great for live work and improvised arrangement. However, Jam has recently been »
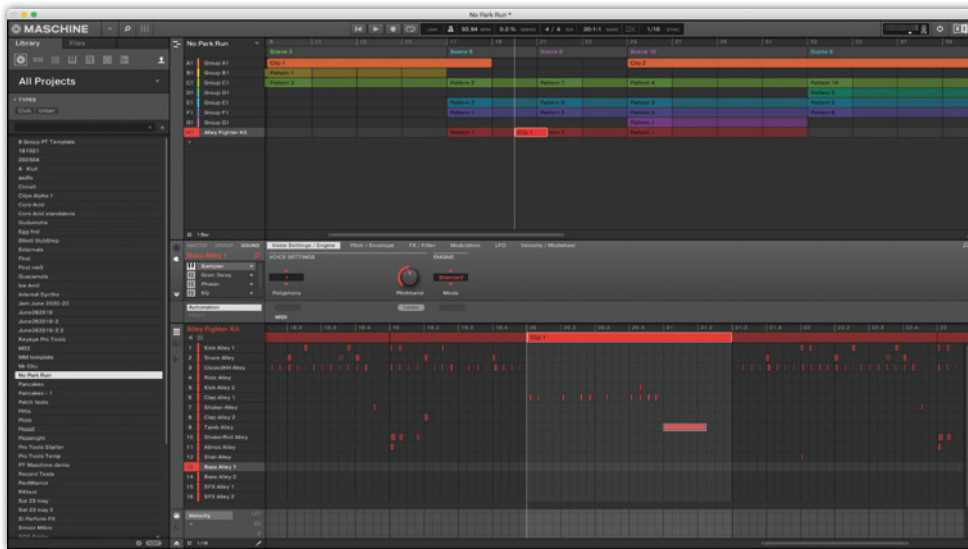
■ Clips as seen in the Maschine software.

discontinued, and NI have not committed to making it work with the Maschine+.

It's not just USB MIDI that's supported on the Maschine+. You can switch to an external USB audio interface. Right now this is only officially supported with the NI Komplete Audio interfaces. These top out at six channels (four analogue), but provide a way to connect mics that require phantom power.

NI have not blocked you from trying other class-compliant USB interfaces. My portable Zoom U-44 showed up as both an audio and MIDI interface. But the real stunner for me was that my Zoom L-12 mixer/interface — which is the hub for all my synths — worked perfectly! This gave me 12 inputs and two stereo output streams back from the Maschine. I could swap from a computer/DAW-centric setup to standalone simply by unplugging my USB hub from the computer and connecting to the back of Maschine+.

So Maschine+ is well connected; it even supports Ableton Link over Wi-Fi.

However, one thing missing compared to some rivals is CV. If you want to hook up to your Eurorack, you're going to need a MIDI/CV converter.

The Big Picture

You could use Maschine+ in place of a computer, and produce and perform music without ever docking with a computer. However, I imagine most users will want to move projects one way or the other. I typically use Maschine to play around with new ideas and jam with hardware synths. Then I'll open the project in the plug-in version to progress the idea in my DAW. Others may go the other way, perhaps to pull apart finished tracks and prep them for live performance.

On the regular, computer-based Maschine the workflow is trivial: you've got one library and one set of connections that you can access whether you're using Maschine as a separate app or as a plug-in. Moving between the untethered Maschine+ and your computer-based setup is not as straightforward as it could be.

To take an idea from standalone to Maschine to your computer, you need to copy across the Project (either via Storage mode or popping the SD card), and make sure any referenced assets are available. Anything you sampled will be in your Projects folder or the global Recordings folder, but anything else you use from your library or user files won't, and there's no equivalent to the software version's 'Save Project with Samples' command. I got pretty used to seeing the Missing Samples dialogue pop up.

Having moved your files, you need to switch to controller mode to continue using

the hardware with Maschine desktop/plug-in. Maschine's on-board audio interface becomes available to your computer, as do the traditional MIDI ports, but USB connections do not. If you had any synths or interfaces connected via USB that you want to continue to access, you'll need to reconnect them to your computer.

The MPCs show how this can be handled much better. When you switch an MPC to controller mode its storage mounts on your computer, so you can simply load your project directly without moving it or collecting assets. What's more, the MPC acts like a MIDI hub so the devices you had connected can be used without reconnecting.

Conclusion

If, like me, you love the way Maschine works, the idea of having it as a self-contained workstation can't fail to appeal. Maschine+ goes way further than I'd imagined, with desktop-class plug-ins and expandable I/O. But do you need it? The regular Maschine gives you all this and more for half the price (assuming you have a computer). But there's obviously a case for live performance. If I was setting up a live show this is what I'd base it around. And then there's the draw of simply having a unit that you can move around, or simply use without the faff of your laptop. It's a luxury item, but priced comparably to Elektron's flagship grooveboxes and the premium MPCs.

Though the feature set exceeded my expectations, I've not fully fallen in love with the Plus yet. There are things that could do with refinement. Old frustrations like the inability to capture Scene or Snapshot triggers into the timeline are joined by new ones, like the audio recording limitations and pitfalls when moving a project to your computer. I hadn't expected the amazing synth portfolio, but having got it I soon wanted more access to them beyond the preset macros. But it's early days, and while there's plenty of room for improvement, the Maschine+ is already incredibly powerful as both a beat workstation and a DAW-free synth hub. And that rich sound palette and fast and fluid workflow always keeps me coming back to Maschine. ▰▰▰

$ $1399
W www.native-instruments.com

LEGEND REDUX.

THE "WHITE CONE" HAS RETURNED!

Avantone Pro and legendary mixer Chris Lord-Alge have already designed the most faithful reproduction of the world's most iconic studio monitor: The **CLA-10**. But they couldn't just stop there: Introducing the **CLA-10A** — an active-powered version of the infamous "black box with the white cone."

The CLA-10A packs an impressive 200WRMS, thanks to it's monolithic Class AB amplifier. And though that may be impressive, we couldn't just stop there...

The CLA-10A's have an all new "**VTPC**" knob or "Variable Tissue Paper Control." This VTPC allows you to simulate any thing from the Vertical "m" model to the horizontal "Studio" model and everything in between. No more debate over the best tissue paper to use — VTPC has you covered!

Black box. White cone. High-range frequency. It's not known as the world's best — it's known for the being the world's best reference — the sound that creates hits.

Arturia KeyStep 37

Controller Keyboard

Arturia's popular KeyStep controller goes up a gear.

SIMON SHERBOURNE

Think of the KeyStep 37 as a MkII deluxe version of the classic KeyStep. (The original 32-key model is still around, providing a lower-priced, more backpackable option). Like its smaller sibling, the 37 is a MIDI keyboard with an arpeggiator and step sequencer. Added to the mix are an extra five keys, MIDI CC controls and advanced chord and scale functions.

Most of the multitude of compact MIDI keyboards on the market focus on controlling DAWs and apps. They typically provide a mix of controls for software mapping, and sometimes add some smart integration with particular software. Nearly all are USB-only. The KeySteps take a unique approach that's more about playing and generating musical ideas. The KeyStep Pro is a central sequencing hub for a hybrid setup, while the 32 and 37 are ideal as companions to synths or other sequencing environments.

First Steps

The KeyStep 37 looks like a slightly elongated version of the KeyStep, because it is. On the rear panel, the connectivity is the same except that the USB connection has been upgraded to a full-sized port, and there's now a power switch. As well as USB there's regular 5-pin DIN MIDI In and Out, a modular voice set of Pitch, Gate and Mod, analogue clock in and out, and a sustain pedal input. The 37 retains the pair of dip-switches for clock settings, making a refreshing change from menu-diving.

Up top, the extended keyboard now has key lights, and there's a new section with four knobs and a button. All the

> ### "The KeyStep 37 goes to places most MIDI controllers fear to step..."

other controls carry over from the 32, including the Sequencer/Arp slider switch and eight-way selector knobs which have a satisfyingly old-school synth feel.

I spent much of the review with the KeyStep plugged directly into an MPC Live via USB. This was able to power the keyboard as well as host it as a class-compliant MIDI device. If you want to run standalone but use the MIDI or CV outputs, you can juice the device from any regular portable charger or add the optional mains PSU. If you're plugged into a computer, MIDI notes sent to the KeyStep 37 via USB goes to all outputs, so it can work like a single-channel MIDI/CV interface.

Arping On

Many controllers and synths feature arpeggiators, but the KeyStep 37's is a little bit special. It goes further than the usual modes and controls you'd expect, with niceties like exclusive and inclusive up/down mode, variable gate length and swing.

Joining the modes from the original 32-key model and KeyStep Pro is Pattern mode, which was an innovation on the MicroFreak synth. This creates a random note pattern that repeats instead of randomising each cycle, and has a specific length that you can set up to 64 notes. You can add more notes to alter the pattern, and change the length on the fly. It's a really great way to generate semi-random sequences within parameters you can control. The only thing you can't do (which you can on the MicroFreak) is capture the result to a slot in the Sequencer.

The Arp and Sequencer both benefit from a new scale option, which contains

In Control

When not controlling chords, the four new knobs on the KeyStep 37's panel generate MIDI CCs, and the Chord button serves a secondary duty of cycling through four control banks. The assignments can be edited directly on the unit, but you have more straightforward and detailed control of this from Arturia's standard MIDI Control Center software.

The knobs can have independent MIDI channels or follow the User channel, and you can set ranges. There's even a knob-scaling option to avoid jumps when changing banks. For CV Mod control you can choose the Mod touchstrip, velocity, or aftertouch as your source.

The MCC app also reveals some advanced features that really impressed me. For example, you can set a port and channel for transposing the sequencer from an external source. And you can choose whether sequence changes are queued, instant and reset, or instant from the same step position. Arturia are always strong on details like this.
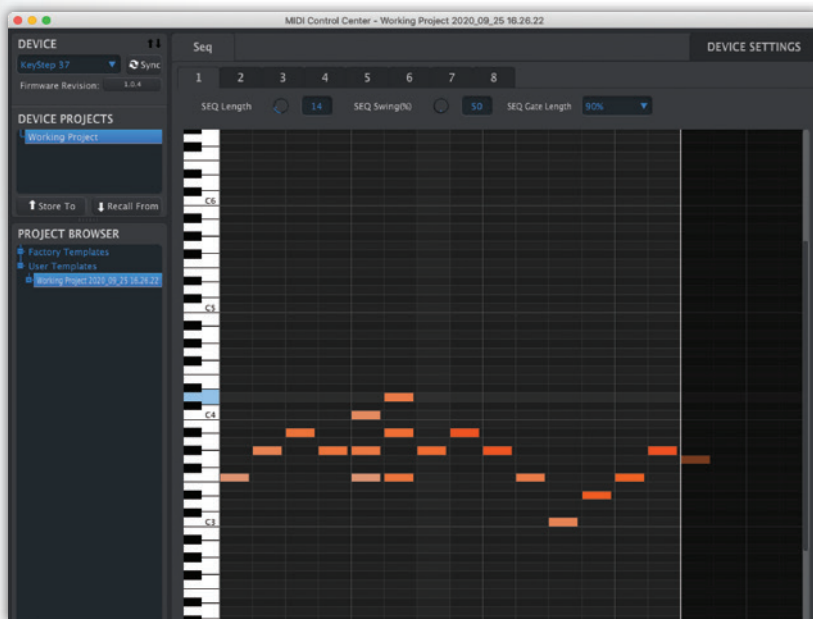
Advanced settings, back-up and control assignments are available in the MIDI Control Center, and it even has a piano-roll sequencer editor.

notes to a chosen mode and key. Other than Chromatic, there are only three preset scales to choose from (Major, Minor and Blues), but you can very quickly define a User scale by holding the notes that you want to include.

Striking A Chord

The original KeyStep 32 has a basic chords feature where you can capture a chord shape that you're holding, then play it with single keys. The 37 can do this and a lot more besides, being able to auto-generate chords of various types, choose the number of notes, and transpose in key within a selected scale. These chords can be combined with the Arpeggiator.

But the most striking and addictive feature of the KeyStep 37 is the Strum function. I've used the strum option on Native Instruments' Maschine controllers, where you manually strum a held chord via a touchstrip. The KeyStep 37's strum works differently: it plays chords as a sequence of notes in ascending or descending order, like strumming a guitar chord. The right-most of the new knobs sets the speed and direction. At the far end of the knob's range are sync'ed strum speeds, providing an alternative take on arpeggiation.

An added expressive feature is being able to tie the number of notes in the chord to velocity. Play softly for smaller chords, and strike the keys for up to 16 notes.

In Sequence

The KeyStep is probably best known for its classic step-sequencing functionality.

The KeyStep 37's sequencer is fundamentally the same as on the classic 32, but it is elevated by the new scales and chord features.

The sequencer primarily uses step-entry mode, although there is a live record/replace mode which quantises to the nearest step. In the classic SH-101 style, you enter note and rests with the transport stopped, and the sequence length is simply determined by the number of entries. At any time, however, you can adjust to a specific loop length. Eight sequence memory slots are available from the first dial.

What makes the KeyStep quite magical is interacting with sequences in real time. You can transpose the sequence with the keyboard and flip between scales. You can overdub or replace notes in real time within the current loop, or use the Append mode to keep adding extra notes to the end of the sequence. This

last is particularly fun, throwing off your loop length in unpredictable, polymetric ways. The enhanced chords functions, including Strum, can also be triggered by the current sequence.

Conclusion

The KeyStep 37 goes to places most MIDI controllers fear to step, with full standalone capability and both MIDI and CV connectivity as well as the usual USB. But more than that it's an inspiring musical companion and idea generator for any synth, sequencing workstation or DAW. ▰▰▰

summary

A great step up from the original KeyStep, with absorbingly fun sequencing and performance features.

$ $169
W   www.arturia.com

PHIL WARD

Ocean Way HR5

Active Monitors

Ocean Way's newest studio speakers bring the sound of '70s horn-loaded mains to the nearfield monitor format.

Who says all nearfield monitors are the same? The subject of this review is the Ocean Way HR5 active nearfield monitor and it's very much not run-of-the-mill, either in terms of its appearance or electro-acoustic design. It's an unusual one there is no doubt. Before I get on to why the HR5 looks the way it does, and what that means, perhaps a little Ocean Way background would be appropriate.

The Ocean Way name has been around in pro audio since the mid-1970s, when teenage musician and audio enthusiast Allen Sides began building speakers and recording bands in a rented garage in Los Angeles. Sides named his business Ocean Way after the street in which the garage was located. Outgrowing the garage, Sides rented space at the United Western Recorders studio on Sunset Boulevard, where his recording and speaker-building business then grew to such an extent that in 1988 he was able to buy the entire studio, and rename it Ocean Way. The Ocean Way studio group grew to four

high-end complexes (Hollywood, Sherman Oaks, St Barths and Nashville), for which Sides designed all the main monitoring, before it was sold in 2013. Sides' Ocean Way business now manufactures a range of both pro-audio monitors and high-end hi-fi speakers.

The HR5 is the latest and the most compact in the Ocean Way range to employ the same horn-loading techniques as the company's main monitors. It's also the only fully horn-loaded monitor of nearfield proportions that I'm aware of (although at 31 x 45 x 37 cm and weighing just over 17kg, it's a relatively big and heavy nearfield monitor).

Within the cabinet, all but hidden behind its complex, multi-profile horns,

are an 18cm bass/mid driver and a 25mm soft-dome tweeter. Substantially obscured as they are, it's not easy to discern much concerning the drivers' construction. Around the back of the cabinet is an amplifier heatsink and connection panel, and an unflared circular reflex port. The choice of an unflared port will lower the volume level at which the airflow will become turbulent, with port compression and distortion effects becoming apparent at lower volume levels than they otherwise would.

The HR5 amplification offers balanced XLR analogue and AES3 digital inputs, an AES3 output, input sensitivity adjustment and four configuration LEDs that illuminate in conjunction with a pinhole

button. Pressing this button cycles through configuration options for input selection, digital input left or right channel assignment, and an 85Hz roll-off option for use when the HR5 is partnered with a subwoofer (the Ocean Way S10A is advertised as a suitable option). Different LEDs illuminate depending on the configuration selected. In terms of power ratings, the HR5's ICEpower Class-D amplification offers a reasonably generous 125 Watts for the bass/mid driver and 90 Watts for the tweeter.

So, behind the plane of the drivers, the HR5 is actually a relatively conventional two-way compact ported speaker. The compression horns located in front of the drivers obviously mark it out as a little different, however, so I'll spend a paragraph or two attempting to get our collective heads around fundamentals of horn-loaded speakers.

Sounding The Horn

Back in the days before transistors, and certainly before Class-D amplifiers made high audio power levels so easily accessible, relatively low-power valve amplifiers were the only option and, as a result, speaker efficiency (electrical power in against acoustic power out) was at a premium. Even now, typical 'naked' bass/mid drivers struggle to achieve single-figure efficiency. And that's where horn loading initially came in. By increasing the effective efficiency of moving-coil drivers, horn loading made

Ocean Way HR5
$3286

PROS
• Dynamic, horn-loaded character.
• Neutral tonal balance across a wide horizontal angle.
• Plays loud without complaint.

CONS
• Amplification a little noisy.
• A little short on reproducing explicit mix detail.
• Restricted vertical dispersion.

SUMMARY
The HR5 measures up to its unusual looks with an unusual performance, especially in terms of its wide horizontal but narrow vertical dispersion. Perhaps not suited to all genres, the things it does well, it does very well indeed.

possible the public address systems in movie theatres and live music venues, and the high volume levels required of main monitors in recording studios. It was only the arrival of transistor power that began to make horns unnecessary.

But how, I hear you ask, does a horn increase efficiency? It's all about radiation impedance. When a naked driver diaphragm moves in a fluid (air, for instance), the power it transfers depends on the fluid's density. A big difference

> "The HR5 can claim a direct lineage back to West Coast rock & roll history… even I couldn't resist playing 'Hotel California' for the first time in many decades."

between the density of the diaphragm and the density of the fluid is equivalent to an impedance mismatch in an electrical circuit. Imagine trying to power yourself along by doing a front crawl swimming stroke in air rather than in water. It doesn't work in air because air is very much less dense than water. The higher density of water stops the molecules from moving sideways to get out of the way of your gracefully sweeping hands. Fish can move rapidly in the water with small fins, while birds need big wings to stay in the air, all for the same reason.

Adding a horn to the front of a speaker works because, down at the business end, otherwise known as the throat, the horn essentially stops the air moving sideways and getting out of the way of the diaphragm. The length and flare of the horn then progressively blend the radiation impedance at the throat into that of free air. The principle is pretty simple, however, as with many simple principles, if you dig a little deeper it gets complicated. The complications arise mostly because we live in a universe in which the laws of physics dictate that the most effective way of transitioning from the throat to an exit in the listening space requires horns many metres long (and consequently many metres in height and width). Usable horns have to end at some practical length, however, and that both puts a limit on the lowest frequency at which they will work, and results in resonant standing waves being set up along their length. Above the cutoff frequency, a series of peaks and troughs in the frequency response arise as a consequence of the standing waves. It's these resonant response

anomalies that are often responsible for horn-loaded speakers having a reputation for a characteristic subjective quality.

The shape of the horn as it expands away from the throat is also critical, and while the simplest geometry (a straight-sided cone) can work, the efficiency gain it provides is minimal at lower frequencies. Exponentially expanding horns can provide suitably increased efficiency gain at lower frequencies, but also demonstrate a 'cliff-edge' low-frequency-cutoff.

To these complications of horn length, mouth size, flare rate and flare shape, you can also add the potential extra efficiency gains provided by a 'compression' design, where the cross-sectional area of the throat is significantly smaller than that of the driver diaphragm. Non-symmetrical arrangements, of the type employed by the HR5, where dissimilar lateral and vertical flare and length are also an option. Clearly, when you decide to design a horn-loaded speaker, there's an awful lot of parameters to juggle.

If that were not enough, along with the efficiency gain that results from horn loading, you also get some control of dispersion. With a naked driver, dispersion is fundamentally defined by the size of the diaphragm, but with a horn, dispersion is defined by the size of the mouth and the angle of flare. Ocean Way, in their marketing material, advertise that the HR5 horizontal dispersion extends to ±50 degrees, a number that roughly approximates the average horizontal flare angle of its horns. Now, ±50 degrees is quite an ambitious claim, but no limits were provided, so we don't know if that's -0dB, -3dB or -XdB of the axial frequency response, or over what bandwidth. It'll be interesting to take some FuzzMeasure data and see. As far as vertical dispersion is concerned, Ocean Way claim ±20 degrees, but again, this is without the benefit of any limits.

Returning to the horn-loading efficiency gain for a moment, the reason that it is potentially of benefit in a nearfield monitor is that it means the drivers have to work far less hard to generate the necessary volume levels than they would in a direct radiating design. All other things being equal (warning: all other things are rarely equal!), this should result in lower levels of »

» distortion and thermal compression, and an ability to play louder. There is one fly in that ointment, however: thanks to the HR5's bass/mid horn being relatively small, its cutoff is around 150 to 200 Hz, and below that frequency the system will tend to behave more like a conventional direct radiator. This means that it's potentially the low-mid/bass band, rather than the higher-frequency horn-assisted band, that will define the HR5's maximum volume limits and distortion behaviour.

Measuring Up

So, to some Fuzzmeasure data. Diagram 1 shows an axial frequency response of the HR5 from 300Hz to 20kHz. The measuring mic was 60cm distant and aligned vertically with the lip between the HR5's two horns. The first thing to note is that actually there are two curves in Diagram 1, one for each of the pair of monitors. Ocean Way claim in the HR5 marketing material to achieve better than ±1dB left/right pair matching, but based on Diagram 1 they actually do rather better than that, which is admirable. The second thing to note is the gentle 'dipped' character of the response shape, with the minimum centred at just above 5kHz. This measured response shape confirms some comments Ocean Way made to me about how the HR5's frequency response is voiced for the most useful real-world balance. The last thing to note about the HR5's axial response is that, although the average trend is admirably flat (intentional balance tweaks aside), it's relatively uneven. This suggests there's multiple minor diffraction and standing-wave phenomena at play throughout the bandwidth.

Diagram 2 shows a repeat of the axial response of Diagram 1 overlaid with a response taken at 50 degrees off-axis horizontally. From 300Hz to 12kHz the HR5 is around -6dB at 50 degrees. Below 300Hz, as is the case with most similarly sized monitors, the HR5's dispersion tends towards omnidirectional, and above 12kHz some horn standing wave or diffraction effects impose a couple of sharp response anomalies. In the grand scheme of things however, the HR5's horizontal off-axis response is impressive. There are few conventional, direct-radiating monitors that achieve anything like linear and well controlled dispersion, especially through the midrange, right out to 50 degrees. Finally on horizontal dispersion, the HR5 response anomalies above 12kHz may
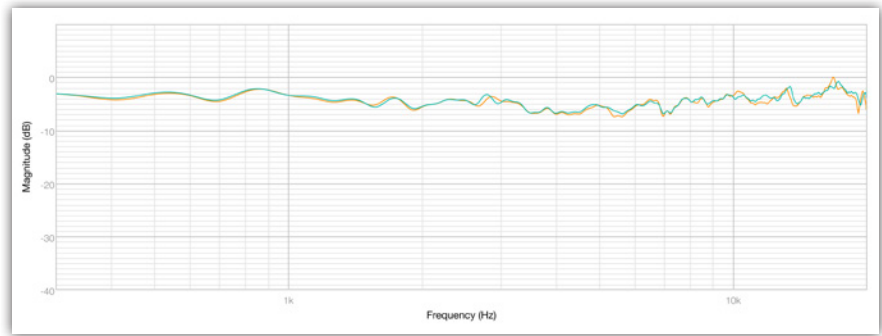
Diagram 1: The orange and green traces show the frequency response matching between two HR5 speakers. Matching is better than ±1dB across their entire bandwidth.
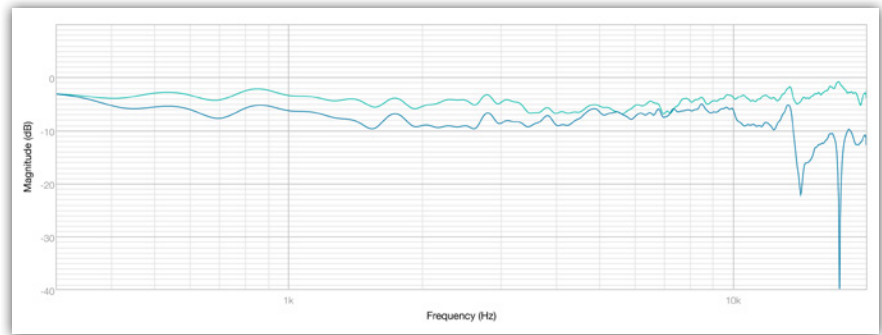
Diagram 2: The response of the HR5, on-axis (green) and 50 degrees horizontally off-axis (blue).

look significant but I'd expect them to be relatively benign in practical terms.

The Ocean Way argument in favour of wide horizontal dispersion is that it helps ensure multiple listeners hear the same tonal balance, and that the stereo 'sweet spot' is expanded. I'm personally not entirely convinced by these arguments. Firstly, nearfield monitoring tends to be, almost by definition, a one-person job, and secondly there's a psycho-acoustic argument that, especially when speakers are used in smaller rooms, directing significant wide-bandwidth audio out sideways to reflect from the side walls does no favours. It will drive the room acoustics harder and bring the reverberant field closer to the nearfield. The extended stereo sweet spot argument is also, to my mind, a little superficial because the perception of stereo is dominated by inter-aural arrival times and not by differences in volume level. Certainly, wider horizontal dispersion will mean that multiple listeners are more likely to perceive a similar tonal balance, but I'm not convinced it can significantly improve stereo imaging (other than by making it less explicit through placing listeners in the reverberant field) at a non-central listening position.

Moving on to vertical dispersion, the Ocean Way claim is ±20 degrees, although again that's without any defining limits.

Diagram 3 illustrates what I measured. At 20 degrees upwards, the HR5 again shows well-managed dispersion that's generally within about 4dB of the reference axial response. However, 20 degrees below shows a significant dip around 1.5kHz (the output above about 5kHz is also attenuated slightly but that's partly a result of the increased distance between tweeter and measuring mic). The dip is almost certainly a result of phase cancellation between the two drivers in the crossover region, where their outputs overlap. A phase cancellation dip in one vertical off-axis direction is very common in multiway speakers, so the HR5 is by no means alone in displaying such a phenomenon, however the relatively low crossover frequency of the HR5 puts the dip at a sensitive point in the bandwidth and I'd expect the dip to be audible. Having said that though, I guess listening to the HR5 from a below-axis position is unlikely.

Diagram 4, the last FuzzMeasure curve before I get to describing my subjective feelings about the HR5, shows a frequency response obtained by placing the measuring mic on the central axis about halfway 'down' the HR5's low/mid driver horn. There's no particularly definitive data to be gleaned from Diagram 4, however I've included it for interest because its uneven nature perhaps illustrates the horn »

Collection 8

Legendary Keyboards Reinvented

It all

started here.

ARTURIA

_The sound explorers

» standing wave phenomena I mentioned earlier and how they introduce resonant dips and bumps. For example, I wonder if the sharp feature just below 200Hz is evidence of the horn cutoff? In any event, a naked driver in a baffle measured with a microphone at a similar distance would almost certainly display far greater evenness of frequency response.

Listening In

I listened to the HR5 in my usual setup with the monitors around 1.5m apart, mounted at seated head height on extremely rigid wall brackets. As usual, the diet was a variety of Pro Tools mix sessions and well-known tracks ripped from CD. The first thing to note was a slightly disappointing audible level of hum and hiss. We give active nearfield monitors a very hard time in terms of amplifier noise through listening so close, and the HR5's noise isn't a deal breaker (and it's by no means the worst I've heard) because it becomes effectively inaudible once music is playing. However, it is audible in the gaps and slightly distracting when nothing is playing.

Moving on to playing some music, The HR5's fundamental tonal balance immediately sounded well judged and to my ears right on the money. I'd have few worries making mix or mastering judgements on overall balance and I'd expect mixes to translate well in those terms.

The second characteristic of the HR5 that I established pretty quickly is that, while it demonstrates good subjective tonal consistency horizontally, it's not so strong vertically. The 1.5kHz dip at 20 degrees below axis was audible and I had to be careful with my posture while listening. If I slouched and let my head drop too far there was a noticeable tonal change; voices in particular became somewhat recessed and dull. Vertical dispersion upwards however was far more consistent and reliable.

So the HR5 scores well subjectively in terms of overall balance and horizontal dispersion, and it displays the dynamic, explicit character I'd expect of a horn-loaded monitor, with apparently no practical limit on loudness (I gave up before it did). If you regularly monitor at high levels, the HR5 will be of interest. Similarly, despite its unflared port design, I didn't have many issues with the way the HR5 plays bass. Subjectively, it didn't seem quite as extended in terms of
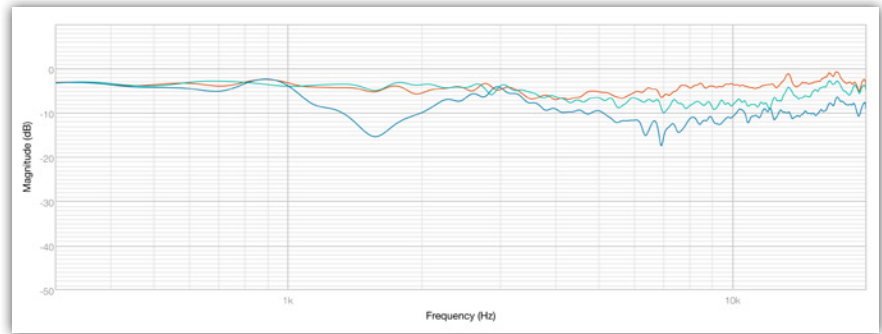
Diagram 3: The HR5's vertical dispersion characteristics. The red, green and blue traces show, respectively, the on-axis, 20 degrees above, and 20 degrees below-axis responses.
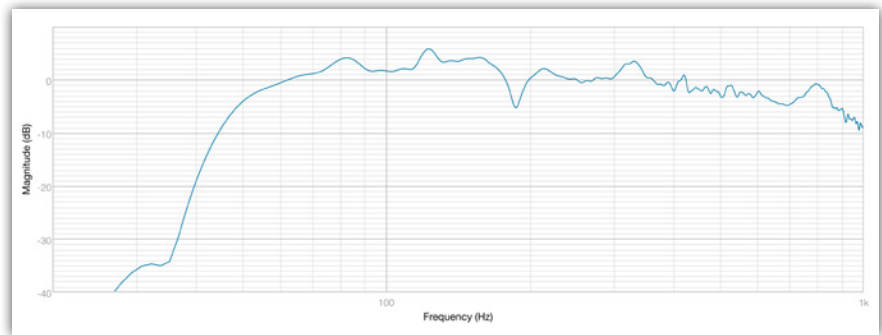
Diagram 4: The HR5's LF response, measured at the lower horn.

bandwidth as its 45Hz specification would suggest, and I suspect a good closed-box monitor would offer more in the way of low-frequency detail, pitch security and timing, but the HR5 bass was relatively free of ported speaker flaws and the overall bass level was well judged.

Almost all monitors are of course a mixed bag, and along with its positives the HR5 displays a couple of subjective negatives for me: the way it presents mix detail, and some coloration on voices. In the first case, I never quite got the feeling that I was able to hear really deeply into a mix. Some mix elements appeared a little veiled: plug-in effects, for example, sometimes seemed a harder to judge than I'd like. Stereo image focus and depth also appeared less explicit than I'd expected, and the sense that some monitors provide, of all the components of a mix individually laid out for inspection, wasn't quite there for me with the HR5. And in the second case, I often use simple voice recordings when appraising monitors and with the HR5 I felt the balance between vowel and consonant elements was slightly awry, with the former suppressed and the latter, especially sibilance, exaggerated. Female voices in particular also occasionally took on a mild nasal character, and cymbals sometimes sounded a little thick.

Despite all that, the HR5 has much to recommend it: its roots are firmly in

traditional, horn-loaded American studio monitoring, and it undoubtedly looks very cool, but more than that, the HR5 can claim a direct lineage back to West Coast rock & roll history. I can absolutely see that for some users this will all be enough, and even I couldn't resist playing 'Hotel California' for the first time in many decades (and, for the record, I don't mind admitting I loved it). To my mind, while the HR5 is perhaps not an ideal monitor for acoustic folk, jazz or traditional classical genres, if I were working in EDM, guitar-based indie or commercial pop production, for example, its wide bandwidth, dynamic nature, well-judged tonal balance and ability to play loud without changing character or showing any signs of strain might well be exactly what's needed.

Monitoring is of course a personal decision and my opinion just one among many. That the HR5 didn't entirely deliver for me doesn't mean it won't for you. If I were in the monitor market at the HR5's price and wanted to try something a bit leftfield as a contrast to all those conventional active two-way boxes, the HR5 would be a perfect fit. **■■■**

$ \quad $3286 per pair.
T \quad Ocean Way Audio +1 818 847 8757
E \quad info@oceanwayaudio.com
W \quad www.oceanwayaudio.com
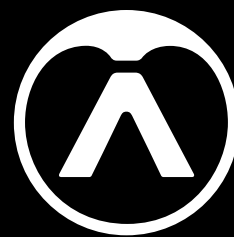
MOD Devices MOD Duo X

Hardware Plug-in Host & Controller

Now you can rock the MOD on your desktop!

RORY DOW

Back in November 2017, Paul White reviewed the MOD Duo, a guitar stompbox that hosts software plug-ins that you can arrange as virtual 'pedalboards'. A key feature is the open platform, which allows third-party developers to release compatible plug-ins.

Over the product's lifetime, the folks at MOD Devices noticed that more and more people were using the MOD Duo as a desktop device, creating not just guitar effects but also synths, MIDI processors and even CV processors. So they decided to make another version of the MOD Duo, this time aimed at those desktop musicians. The result is the MOD Duo X.

Like its predecessor, the MOD Duo X is a stereo/dual-mono device that can host open-source Linux LV2 plug-ins. Editing is done via a web browser, with the MOD Duo X connected to your computer. The editor gives you a sandbox playground for creating modular pedalboards, as well as assigning the MOD Duo X's physical buttons and knobs to various parameters. There are hundreds of plug-ins to choose from and there's also a large library of user-made pedalboards. Once you're done editing, your virtual pedalboards are

available on the device itself, to be used completely standalone.

Hardware Mods

The unit measures 18 x 16 x 4.5 cm and it feels very solid. There are twin mono quarter-inch audio inputs and outputs (with true bypass circuit) on the right and left sides, along with a headphone output and expression pedal CV input. The rear is home to a 12V power input, Control Chain Ethernet port, USB type-B socket for connecting to your computer, a USB type-A port, which allows for hosting USB-MIDI peripherals, MIDI In and Out DIN connectors and an S/PDIF digital output on eight-inch jack, which mirrors the analogue outputs.

The Ethernet socket is a proprietary connection for MOD Devices' Control Chain protocol. You can use it to connect their MOD Footswitch device, which gives you four footswitches that can be configured to do anything. The Footswitch has two LCD screens which show customisable labels for each switch. Control Chain devices can be daisy-chained so you can add multiple Footswitches or other compatible devices. The software and hardware for the MOD Footswitch and the Control Chain protocol are, of course, open source — so DIYers or other manufacturers can make their own compatible devices.

The Duo X's two monochrome, backlit LCD screens on the front panel are placed neatly above two groups of knobs and buttons. Because they are LCD and not OLED, they can be a little difficult to see unless you're directly above them, but they do the job. Their main function is to show you what parameters the knobs are assigned to and their current value. The

MOD Devices
MOD Duo X

$749

PROS
• Very well built and reliable.
• Mature and stable software.
• Open-source platform.
• Hundreds of plug-ins to play with.

CONS
• None of significance.

SUMMARY
The Mod Devices open-source platform gains a new desktop flagship hosting device, capable of running hundreds of Linux LV2 plug-ins and arranging them in modular-style virtual pedalboards for standalone use. It should appeal to anyone who likes to 'roll their own'.

layout on the screens matches the physical layout, so it's all very intuitive. There are two large clickable, endless, rotary encoders, eight potentiometers and four backlit on/off buttons, all of which can be assigned to multiple parameters with variable ranges.

Software Mods

Whilst the MOD Duo X comes installed with a variety of 'pedalboards' to get you started, I imagine most users will be keen to roll up their sleeves and build their own. Connect to your computer via USB or Bluetooth and enter the device's IP number into your browser, and you're presented with the pedalboard editor. All hardware inputs are on the left and all outputs on the right; these include all audio, MIDI and CV connections. Then, it's as easy as adding a few plug-ins and connecting them together using virtual cables. Each plug-in has its own graphical interface and the whole system looks great and feels very slick.

Every plug-in has an info page with details about its author and its features, and you can even see what pedalboards it has been used in. Plug-ins are categorised, which helps greatly when there are so many. Not all plug-ins are installed on the unit by default. The review model we received had around 80 plug-ins already installed, but following the link to the plug-in store revealed more than 200 more which could be installed and used.

One of the most important aspects of building your own pedalboard is assigning the physical controls. Although the MOD Duo X has 10 knobs and four buttons, there are actually three pages which can be assigned, giving you up to 42 assignable controls per pedalboard — and that's before you add MIDI, CV or Control Chain assignments (see the Control Chain box). Once you assign more than one page of controls, a single button on the front panel is used to advance to the next page, and the displays update accordingly.

Assigning controls to parameters within the pedalboard is easy. You choose which plug-in to work with and click the settings cog just above it. This opens up a modal dialogue box in which you can assign any physical device control, MIDI control change, CV input or attached Control Chain device to control that parameter. You can also give the parameter a name which will appear on the LCD screen and select a value range. Additionally, each plug-in can have presets, and you can assign a control to scroll through those, which is a good way to make more drastic changes. Once you've

assigned a knob or button to a parameter it is greyed out and unavailable to assign elsewhere. It's also possible to create macro assignments by enabling the output of a 'MOD Control to CV' plug-in as a mapping source; a single CV signal can be mapped to as many parameters as desired.

Pedalboard sizes are only restricted by CPU and RAM and there are live meters for both at the bottom of the screen, so you can always see where you're up to. The MOD Duo X is based around a beefy ARM Cortex 64-bit Hex-core processor, so there should be plenty of power for most projects.

As soon as a pedalboard is saved, it is stored on the unit along with any plug-ins needed, and can then be used completely standalone. If you choose to load pedalboards from other users, any plug-ins not already present will be installed. Pedalboards are organised in banks to help keep things organised and you can browse banks and load pedalboards directly from the unit. You can also use MIDI Program Change or an attached Control Chain Footswitch to switch pedalboards.

At any point whilst performing with a pedalboard, you can save a snapshot. There are two dedicated snapshot buttons on the front panel and you simply long-press so save a snapshot, and short-press to reload it. This can be really handy during a performance, as it allows you to return to a specific state quickly.

Being a plug-in host, the MOD Duo X must also deal with tempo and it does so rather well. You can choose between setting an internal tempo, or sync'ing to MIDI clock or Ableton Link. Tempo (internal), time signature and transport controls can all be assigned to front-panel controls.

Make Your Own Plug-ins

For those that code, one of the MOD Duo X's biggest selling points will be the ability to make your own plug-ins. Max/MSP ❯❯

The pedalboard editor makes connecting plug-ins easy with drag-and-drop virtual cabling.

integration is probably the easiest route in. MOD Devices have worked with Max/MSP's creators Cycling '74 to make a Max package that can take code exported with Max/MSP's gen~ object and compile it to an LV2 plug-in, ready for use in the MOD Duo X. Several of the library's plug-ins were created this way.

Of course, Max/MSP isn't the only way to code plug-ins. Higher-level languages like Pure Data, Faust, C or C++ can also be used to create LV2 plug-ins. The MOD Devices website, wiki and forums have plenty of information and tutorials on how to go about creating your own plug-ins.

Conclusion

There are several aspects of the MOD Duo X to consider. Firstly, the unit itself is solidly built and even comes in its own carry case. The knobs feel sturdy, the buttons are nice, and the audio and MIDI connections are all on grown-up, full-size sockets. Overall, this is a very well-built device and I have no doubt that it would survive the punishments of being gigged.

Then there is the computer side of things. MOD Devices have put years of work into their pedalboard editor, platform and plug-in library, and it shows. The editor is simplicity itself to use. Any reservations I had about working in a browser were quickly batted away by the slick and

Pedalboards are not just for guitar effects. Here's a drum machine taken from the pedalboards library.
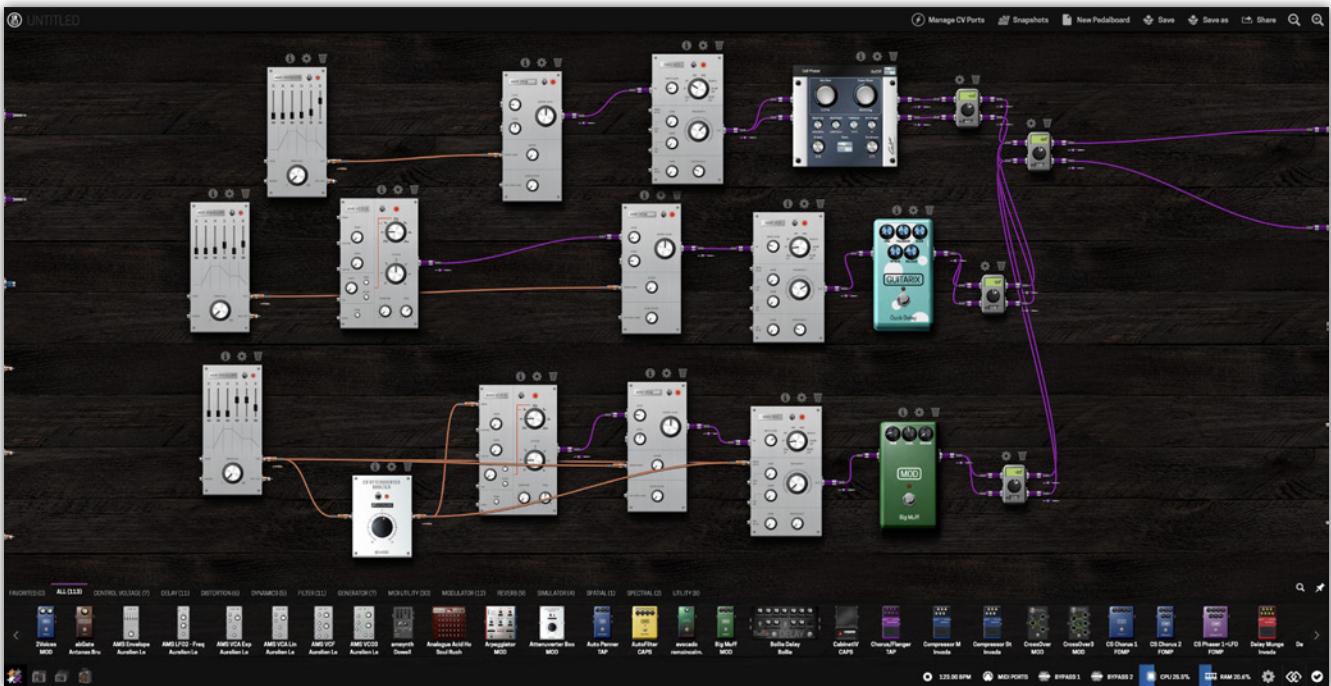
very good-looking modular playground; it was a pleasure to use. I never experienced any disconnections, glitches or difficulties — another thumbs up.

Thirdly, there is the standalone operation. Once again, the unit is simple to use and works as intended. All the pedalboards I tried worked great in standalone mode, the system seems rock solid and stable. Being able to decide exactly how the hardware responds to knob-tweaks means you can build effects and generators that suit your live needs exactly. I wouldn't hesitate to use it in a live situation.

Finally, there is sound quality. This is much harder to analyse, mainly because it's so dependant on the plug-ins you use, and there are hundreds of them. The sheer number available means you can almost always find something that suits your needs. Some plug-ins are based on quite old open-source code, like the MDA plug-ins which have been around for nearly two decades. But there are plenty of new additions to the library from well-respected developers, and the list is constantly growing. The fact that you can code your own plug-ins will be a major drawing point for some, and there is an active community on the MOD Devices forums to help you with such endeavours.

The MOD Devices platform is both impressive and stable, and the MOD Duo X really is a worthy flagship device in their range. It's hard not to be impressed! While hosting software plug-ins in a hardware box won't be something that everybody has a need for, for those who do, and particularly for those who create their own plug-ins or can take advantage of the project's open-source status, the MOD Duo X should prove a very compelling package indeed. ▰▰▰

$ $749.
T MOD Devices +49 (0)30 9451 2477.
E sales@moddevices.com
W www.moddevices.com

Teenage Engineering
Pocket Operator Modular

Modular Synthesizers & Sequencer

True to form, Teenage Engineering's approach to modular
synthesis is about as idiosyncratic as it gets.

ROBIN VINCENT

The Pocket Operator Modular range was first introduced to the world around the time of NAMM 2019. They were bold, striking, unexpected and slightly odd. With everyone around them going mad for Eurorack, Teenage Engineering seemed to offer a more unique approach with these stunning objects of sound and colour. It's an approach that seems to divide opinion and generates strong discussions about the relative merits of design, aesthetics and functionality. However, there were some issues highlighted by early reviewers in the first run of the POM 400 (the big yellow one), and the POM 170 (the red one) ran into some production problems causing very long delays. Eighteen months after their initial release the dust has settled, they've

had a few course corrections in the latest production run and they are now available through regular distribution channels. So, I thought it would be the right time to dig under the surface of these design statements of music, metal and plastic and see how engaging they could be.

Meet The Family

There are three products within the Pocket Operator Modular or 'POM' range. We have the 16 sequencer, the red 170 single oscillator modular synth and the three-oscillator expanded 400 modular synth. Just to be clear the 170 and 400 are modular synths in the sense that they are made up of individual synthesizer modules that have to be physically patched together. However, as the front panels are fixed the actual modules are not interchangeable so you can't really call it fully modular because you can't change the architecture of the synth. So could they be described as fixed synthesizers with modular articulation? Possibly, but there does tend to be a bit of confusion around the way Teenage Engineering labels and defines things.

The best way to describe the 170 is as a modular TB-303 with a massively overspec'ed sequencer. In its rather gorgeous red console-like form the top half has a single square-wave oscillator, a filter, ADSR envelope, VCA and an LFO. The bottom half integrates the sequencer and touch keyboard from the 16 which is capable of sequencing and gating three more channels of CV than the 170 possesses.

The 400 appears to be designed much more like the ARP 2600, which is the sort of comparison that I'm sure many people would object to, but this realisation made a lot of sense to me in trying to articulate what this synth is all about. It has three oscillators, a mixer, two envelops and VCAs, a filter, a noise generator, a sample and hold, an LFO and a much simpler sequencer. It stands up with a near-vertical rake and will

probably be the most eye-catching thing in your room. Some people have worried about the stability when patching, which I haven't found to be a problem, but I've also seen that a few people have mounted it upside down and laid it flat to give it more of a console feel.

Both synths have a small speaker built in and can run on batteries for perfect portability. Or you can add an optional DC power supply for a more permanent installation.

The physical presence of these two objects is quite remarkable. None of the photos prepared me for how fantastic they are in the flesh. They are beautiful works of electronic art that would look outstanding on your sideboard or coffee table. The bouncy, rubberised patch cables contribute to that and they don't even have to make any noise; you just want to touch them.

DIY

The synthesizers come flat-packed and part of the deal is that you have to put them together yourself. This plays into the idea that these are a bit toy-like, which is a very common comment I hear when talking about them online and is usually meant dismissively. They do have a flavour of Meccano and Lego about them and I believe that's deliberate simply because it's fun and playful. The knobs are, in fact, compatible with Lego Technic pieces, which is very handy for reasons I'll come on to. It comes with a brilliantly illustrated Lego-style manual to guide you through the building process. I built them with my kids, aged 8, 10 and 13, who had no problem following the instructions for the most part.

Here it comes a little unstuck because the manual can be confusing in places. The manual contains the instructions for all three products and appears to be in three separate sections. We built the 170 first and found that a lot of details were missing and had to search around online to find some answers. We ended up having to take the

---

Teenage Engineering
Pocket Operator Modular
16, 170 & 400

$239, $479 & $599

PROS
- Beautiful electronic objects.
- Genuinely modular music-making.
- CV and socket compatible with Eurorack.
- 400 is fantastic value.
- 16 sequencer is immensely powerful.
- Completely portable.
- First run issues with build and tuning have been solved.

CONS
- Can be odd.
- CV range can be limited.
- Those knobs.
- Might be too quirky for some.
- Challenging for a beginner.

SUMMARY
The startlingly designed Pocket Operator Modular synthesizers from Teenage Engineering offer a quirky dose of colour and playfulness with a sometimes challenging modular workflow and imperfect architecture. That sequencer, though.

first thing you try, but wheels are just too big. After experimenting with a few options I settled on what is called Lego Technic 'Bushings' (part #3713 / 6590). They come in a range of colours, cost 2p each and are about 7mm tall and wide and fit over the knobs perfectly. Use a permanent marker to indicate the position on them. This makes the knobs a great deal more comfortable and easier to see.

The Background

When it comes to the sound and architecture of the synthesizers they find themselves both beautiful and wayward, powerful and limited, simplistic and yet complex. They are not one thing or the other, which makes them troublesome to pigeonhole. The 170 makes for a great bassline synth but it's also terrific at FM and Ring Modulation-type sounds and that sequencer can run a whole army of synths. The 400, with its dual envelopes and VCAs, can pull off two synths with detuning, complex modulations and FM all over the place, driven by a weird little sequencer and a frantic heartbeat.

The synthesizer modules were designed by Jon Nensén, also known as The Human Comparator. If you check out his other work you'll find things like the Two Thousand Six Hundred ARP 2600 clone, some interesting Buchla modules and the forthcoming two oscillator Recursive Machine synthesizer. My only point is that this is not a designer of entry-level synths for beginners. He's managed to squeeze a tremendous amount of tone and complex possibilities out of an inevitably compromised budget modular system. As with any modular system it takes time and patience to pull sounds from it, unlike the self-playing semi-modulars that they are often compared to. And so while the price makes it attractive to beginners the workflow can be quite challenging.

The price is very important to keep in mind. The 400 is only $599 for a 13-module modular system in a battery-powered case with its own speaker. Building something similar with basic Doepfer modules comes in at over a grand and you haven't bought the case and power supply yet. A better comparison is with AE Modular where their Starter Rack 2 would have a similar range of modules for a similar price. AE Modular has the advantage of being fully modular but is

thing apart multiple times and struggled endlessly with getting the ribbon cable from the touch-keyboard in place. When we came to build the 400 we discovered to our annoyance that all the missing 170 detail was right there in the 400 section.

One of the criticisms I've read of the build was that the standoffs were not threaded, and so to screw the modules in you had to exert an impressive amount of force onto the tiny screw going into a tiny object with a tiny screwdriver. This is one of the improvements they've made with the production run and I am happy to report that they all now have screw-threads. All except one. We had one module that had the old-style standoffs and I can see how it could make people angry.

The final act of the build is to place the knobs. Once pushed in they are easily as

sturdy as any of the smaller knobs you find in Eurorack. There are two different types for no real reason other than to look interesting and either can fit into the same pot. The larger knobs are great with something like a small paddle sticking out, which makes it easier to turn and usefully indicates the position. The smaller knobs, I confess, I've come to loathe over my time with them. To begin with, you put up with them and blame the modules for being inaccurate or your fingers for being too fat. They do have a tiny indent to show position but I can never see it. After working with these synths for a couple of months I can confidently say that they are the most fiddly and uncomfortable knobs I've ever had to play with. However, there's a solution — Lego. They are like the axle parts of Lego Technic that go into the wheels, which is the

■ Pocket Operator Modular 170 synthesizer.
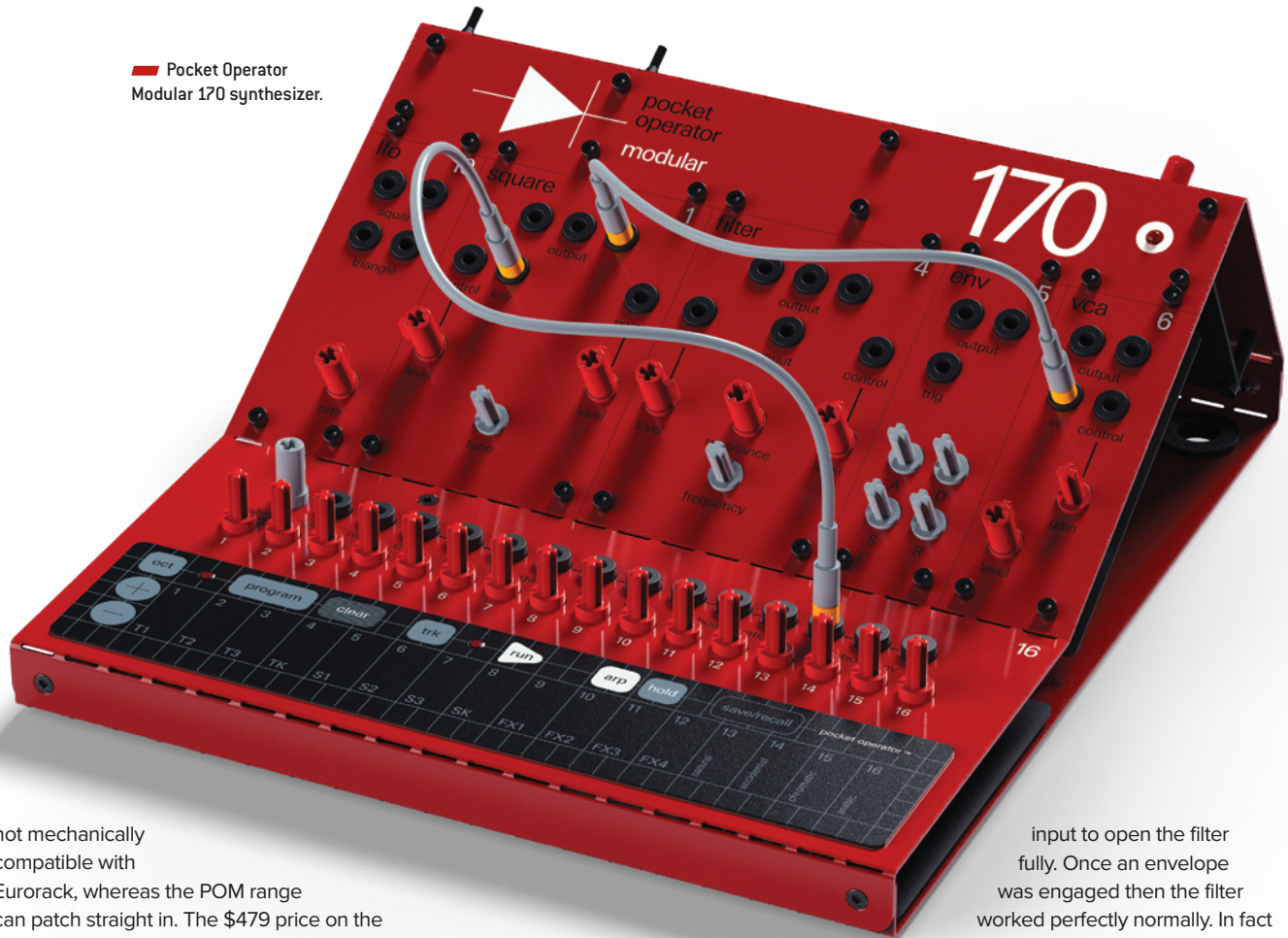
not mechanically compatible with Eurorack, whereas the POM range can patch straight in. The $479 price on the 170 seems expensive when compared to the 400 but that's because it has the $239 16 sequencer built in. I think this is the cause of some consumer confusion as the power of the sequencer is under-appreciated but could also be seen as unnecessary in a single-oscillator synth. If it was $250 with a much simpler sequencer, like in the 400, I think they'd be far fewer objections.

The Synths

When working with the synths you do find character defects that are perhaps implicit in the price but you also discover multiple personalities and surprises that push you into rethinking your approach.

The oscillators generate a single waveform, which is a limitation, but they all have three outputs which can make for an overload of rather delicious audio-rate modulation. The tuning troubles of the first batch are gone and I can track seven to eight octaves without any difficulty. The tuning knobs are not great and require a fine touch to bring the three together. Adding bigger Lego pieces helps with this enormously. The Sine and Saw oscillators both have attenuatable FM inputs which are a load of fun for clangy, spacey and radiophonic-type sounds. The Square

oscillator, which appears on both the 170 and 400, has slightly odd Pulse-width Modulation. The waveform modulates from square wave to pulse from the seven o'clock to 12 o'clock position and then vanishes beyond that. Teenage Engineering say this is by design so that you can throw in some alternative rhythmic gating by pushing a high voltage into that input. That does work quite nicely, actually.

The LFO offers triangle and square waveforms, two of each on a single rate knob. The range of the rate knob is again quite odd in that you only get a small turn before it's up into audio rate. There's no CV control on the rate/pitch of the LFO but you could easily add it in as a drone or use it for some extra bags of that chaotic audio-rate modulation.

There's also something odd about the filter. The cutoff knob doesn't open the filter very far. The 170 and 400 have the same filter module, but I found that you could hardly hear anything of the 170's oscillator through the filter whereas it appeared to open further on the 400. In either case you needed to add an envelope or other modulator, or CV source to the cutoff's CV

input to open the filter fully. Once an envelope was engaged then the filter worked perfectly normally. In fact there are two cutoff control inputs which let you create some interesting filter offsets and modulations. As with the oscillators, there are three outputs on the filter which gives you all sorts of audio processing possibilities if you were patching out to a larger system.

To complete, the overall oddness of the 400 16-step Sequencer is also slightly strange. Before adding the Bushings and permanent pen it's annoying and tricky, but once that's sorted out it's actually quite clever. Each knob sets an unquantised CV value that can be patched to all three oscillators simultaneously. The tempo runs from 'fast' to 'really fast' unless you patch something more sedate into the Clock In. There's no gate output, but you can use the Clock output to trigger your envelopes for every step, or there's a 'PO' Pocket Operator output which runs at a quarter speed and is intended to sync-up one of their little drum machines. Turning a step's knob all the way clockwise will reset the sequence back to the first step, which is an interesting way of setting the sequence length, although until you read about that in the manual it's awfully mysterious. You can change direction with a gate input and there are also inputs for resetting the »
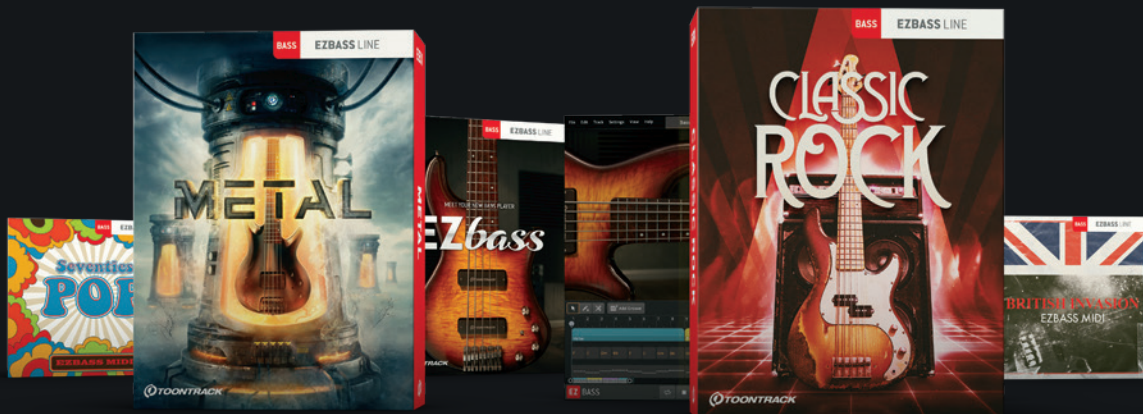
THE EZBASS FAMILY IS GROWING.

Since **EZbass** was released in May, we've already had the pleasure of introducing two EBX expansions and two MIDI packs to the family. Welcome to the ever-growing world of EZbass. Audio demos, videos and more on www.toontrack.com.

WWW.TOONTRACK.COM

TOONTRACK

Pocket Operator Modular 16 sequencer.

sequence, and four others which will send the sequence to step 2, 4, 8 or 12. There's nothing in the 400 itself that can really utilise these sequence mangling inputs. However, once you've slowed it down from an external clock and brought some gates and triggers in from outside then it can get rather interesting.

The other modules act largely as expected. Features like dual outputs on the envelopes are well thought out and the 'Saw' output on the Noise generator makes for a decent source for the 'Rand' sample & hold module. Just don't ask it to sample & hold a tune — it's not terribly stable.

The Sequencer

Finally, we need to look at the 16 Sequencer, which forms half of the 170 and is also available as a standalone, battery-powered, pocket-sized, desktop four-channel CV and MIDI sequencer.

Following the overall theme of the Pocket Operator Modular series the 16 is both brilliant and slightly awkward. The feature set is pretty fabulous. Four independent tracks of up to 64 steps of sequencing with rests, ties and individual pattern lengths. Step time or real-time note entry quantised to one of four scales or completely custom pitches. Individual tempo divisions per track, four swing modes, 16 arpeggiator patterns, muting and soloing tracks. There are performance effects that mess about with gate lengths or drop in glide and legato notes. Track 4 even has three CV and Gate outputs to use with multiple oscillators or to route out as modulations. Save the lot into 1 of 16 slots

for instant recall. Did I mention that it will do all of that over MIDI as well with each track on a separate MIDI channel?

The only glitch in this otherwise fruity matrix of sequencing heaven is the touch-keyboard interface. You have to lean into it quite hard to get it to work to the point that after a while your fingers start to hurt. This is compounded by the fact that you often have to hold a button down while pressing other buttons and you find it's not working because you're not pressing down hard enough. However, once you've got the hang of it the fun and depth of this sequencer overrides the interface irritations. I'd love to be able to plug in a MIDI keyboard, though.

The row of 16 knobs seems mysterious at first. You assume that they set the pitch of your first 16 steps, when in fact these knobs tie in directly to the 16 notes on the keyboard. You could enter all 16 notes in a row and then the knobs do indeed control the pitch of each note in turn. Maybe that's what they are intended for. The manual seems to suggest that they are CV generators to be used to set up modulation tracks rather than notes, but either case can work. You can reset the notes by selecting one of the inbuilt scales and then you'll have to turn the knobs through the current value to capture control again. A bushing and a pen mark dramatically improve the use of the knobs.

Conclusions

I've had these machines a couple of months now and I find them thoroughly entertaining. Sure, they have some

character defects and some personality flaws that occasionally let them down, but considering what's on offer these quirks are easily forgiven. With the 400 you're getting an entirely modular synthesizer experience for under $600. A modular system needs to be learnt and discovered and the 400 is no different. It's playful and portable and begging to be used outside and in unusual places. I'm currently testing the battery life and have it running a sequence on the other side of the room and that little speaker isn't half bad. It can find a whole other life when it gets its hooks into other bits of modular and runs through a decent sound system. The battery life of the 400, by the way, was around four hours on eight AA batteries.

The 170 does seem expensive for a single-oscillator mini-synth, but it has a very under-appreciated sequencer which makes up half the price. I think a 170 'lite' with maybe just a touchstrip and arpeggiator for around $250 would fly off the shelves. However, that sequencer is a really interesting little box with four tracks of CV/MIDI and a ton of possibilities.

There's no denying the toy-like qualities, and it does seem slightly ridiculous that they are improved with some Lego modifications, but a bit of colour and fun are worthwhile attributes these days. While the design and limitations won't please everyone there's a lot to like and enjoy in these quirky bits of electronic playfulness. ▰▰▰

$ POM 16 $239, 170 $479, 400 $599.
W teenage.engineering

SOUND ON SOUND

PODCASTS

Mat Gendreau
5 Top Tips for Parallel Processing

In this episode, Canadian Producer and Educator Mat Gendreau shares his 5 top tips for using parallel processing techniques on voice, bass and drums to help you get a professional polish and more energy in your overall mix.

ADAM Audio - Behind The Brand
20 Years Of Innovation

After a crisis year in 2015, ADAM Audio have come back bigger, better and stronger, with a new US business model and three compelling ranges of studio monitors. Adam Sheppard of ADAM chats to Sam Inglis about the distinctive technology at the heart of ADAM's loudspeakers.

100 Years Of The Theremin
Pioneering instrument reborn for the 21st Century

A special podcast in honour of the 100th birthday. We speak to Cyril Lance (Chief Engineer,Moog), Dorit Chrysler (composer/New York Theremin Society), Bruce Woolley (songwriter/Radio Science Orchestra) and Katia Isakoff (singer/songwriter and producer).

Dean James Barratt
Mix Engineer

Dean James Barratt is a sought-after Mix Engineer based out of The Dairy Studios in London. His recent credits include Shy FX, Joy Crookes, Ray Black, James Vickery, Rag n Bone Man, Craig David and Jess Glynne. Dean talks to Sam Inglis about his personal road to becoming a freelance Mix Engineer, what he thinks defines that role and tips on making it.

Follow our channels by subscribing to the shows on Apple Podcasts, Google Podcasts, Spotify or wherever you get your podcasts.

Listen on **Apple Podcasts**   Listen on **Google Podcasts**   LISTEN ON **Spotify**

All shows are mastered to the highest quality the podcast channel will support and are in stereo.

Check out our website page for further details

www.soundonsound.com/podcasts

Access Analog
Remote Outboard Processing

Love the sound of high-end analogue kit, but don't have the space or the money to buy it? No problem...

SAM INGLIS

Access Analog is both a business name and a mission statement. The goal is simple: to make high-end studio equipment available to everyone who would like to use it. Perhaps you'd love to mix through analogue outboard, but can't afford it. Maybe you work on the road and you can't always be where your hardware is. Or possibly you've reluctantly given up on outboard because of the recall issues. Either way, Access Analog seems to have something to offer.

The service is enabled by two technological innovations. One is a free, cross-platform plug-in that sends audio from your DAW to a remote server for processing. The other is a system of robotics, which makes analogue studio hardware both recallable and automatable. Together, they allow studio gear in another location to be accessed directly from your DAW's insert points. If you know you're going to need a particular piece of gear at a particular time, you can book it in advance in half-hour slots; alternatively, you can simply buy Access Analog credits and use them when inspiration strikes, as long as someone else hasn't booked out the gear you need.

Enter The Matrix

The plug-in is called Analog Matrix and, operationally, it's not unlike using a conventional plug-in 'chainer'. On the left is a graphical menu showing the available equipment. This reflects the selection you make in the Rack pop-up, where you can choose from different collections of equipment in different physical locations. Access Analog also refer to these locations as servers, and each instance of the Analog Matrix plug-in can talk to one server at once. At launch there was only a single server available, but more are in the works, and I was able to preview a mouthwatering selection of gear that's about to come online in the new Nashville server.

Mouse over an item in the list and you'll see a helpful tooltip telling you when it's reserved, and how much time one Access Analog credit will buy you on that item. This varies from about 4.5 up to 8 minutes per credit, and only kicks in after 45 seconds, giving you the chance to quickly audition things without cost. The core server also offers a couple of Warm Audio processors that are permanently free, so you needn't waste paid-for time learning

how it all works. In addition, Credits are sold in packages from 15 to 400; there is a discount for buying in bulk, so although the 15- and 50-credit packages cost $15 and $50 respectively, the 400-credit package is only $320.

To use these tasty bits of hardware, you simply drag them onto an empty slot in the upper section of the main window. The plug-in itself is available in mono, stereo and mono-to-stereo versions, but always shows a stereo signal path here to reflect the fact that some of the gear is inherently stereo. Some of the mono processors are also available in multiples, and if for example you use two Pultec EQs or UA 1176 compressors, you can click a chain icon to link their controls. However, side-chain linking is not currently available in this scenario.

There are five slots in total, and equipment can be reordered simply by dragging and dropping. Each slot that you use introduces a stage of A-D and D-A conversion — there's no way to route the output of one unit into the input of the next in the analogue domain — but the rack uses high-quality converters from Antelope Audio and I never noticed any degradation in sound quality, even when chaining three or four processors. Each slot has its own ±12dB gain trim control, as well as a Mix control for parallel processing.

The piece of gear currently selected for editing is highlighted with a green rectangle, and the actual editing is carried out through a (somewhat) photo-realistic representation of the front panel. It's little different from working with a plug-in emulation of the same piece of kit, except that there is a lag between making an adjustment and hearing the results.

There are one or two operational quirks associated with Analog Matrix, though you get accustomed to them soon enough. One is that although there are factory and user presets associated with individual pieces of equipment, it's currently not possible to store and recall entire chains. This wouldn't be an issue except that it's

also not possible to move the plug-in from one channel to another without losing your settings, and settings are not remembered when you save and reload a DAW project. Some way of recalling chains of processors is on the company's road map for future development.

Return To Sender

If you have a super-fast broadband connection, you can send uncompressed 24-bit audio directly to the server in real time. You may then be able to use Analog Matrix 'live' at the mix, exactly as you would a hardware insert within your DAW. That, however, depends on your DAW being able to compensate for the delay that Analog Matrix introduces into the signal path. By default, this is 1800ms, which is way higher than Pro Tools can manage, but other DAWs are more accommodating. Pro Tools users can work around this to some extent by routing everything that's not being processed by Analog Matrix to an aux and using another instance in bypass on that bus, but it's a bit of a kludge.

Those of us who live too far from the information superhighway will need to use Analog Matrix in data-compressed mode. This allows you to use a variety of lossless and lossy streaming options to audition the settings you've made on the remote gear, but obviously you don't want to pay to use high-end analogue equipment only to have the signal compromised by a lossy codec. So, once you've got the

settings as you want them, you can upload the uncompressed audio and have it processed offline (albeit, obviously, not faster than real time), before downloading the results. This can be done either by hitting Play on your DAW and capturing audio into a buffer, or by simply pointing the plug-in at an audio file on your drive. The latter option is faster and supports the »

Access Analog

PROS
• Offers universal, easy access to the kind of high-end outboard most of us will never own.
• Well thought-out and intuitive user interface.
• Massive potential for development.
• Usable even with a mediocre Internet connection.
• Free to try and can be tailored to any budget.

CONS
• It's only possible to store and recall presets for individual processors, not the entire plug-in.
• Introduces more delay than some DAWs can compensate for.
• One or two stability issues in the review system.

SUMMARY
Access Analog does exactly what it says on the tin, providing an easy, affordable way to work with high-end outboard from the comfort of your own studio. You might be surprised at how much difference it can make to your mixes!

It's analogue, Jim, but not as we know it: Access Analog's main server in Colorado is stuffed with classic gear under robot control!

upload of multiple files for batch processing, which is perfect if, for example, you have a ton of stacked backing vocals that you want to process in one go.

What's In The Box?

At the time of writing, the original Access Analog server offers a total of 20 processing devices. These include studio staples such as the aforementioned pairs of 1176s and Pultecs, an SSL bus compressor, a pair of Empirical Labs Distressors, an LA2A, a Manley Variable Mu, a Neve 33609N and an API 5500 equaliser, but there are also some more modern and unusual processors, notably an SSL Fusion, Black Box HG2, Louder Than Liftoff Silver Bullet, Iron Age Audioworks LH95, a pair of Daking compressors and, my personal favourites, a pair of Highland Dynamics BG2 compressors. It's probably fair to say that if you can't make something sound good with that lot, it'll never sound good. However, it's also fair to point out that there are relatively few EQ options in the collection as it stands. It would be nice to have a few more really high-quality equalisers such as the classic Neve and SSL designs. Also missing from the line-up as it stands are effects devices and, especially, reverbs. Even studios that already have a decent selection of high-end outboard might see value in accessing a really good plate or chamber online!

However, the original server only represents the first step in an ambitious plan. High-end outboard represents a massive investment, but at any given time, most of the processors in most of the world's studios probably aren't being used. Having demonstrated that their system works, Access Analog are hoping that other studios will be interested in setting up servers of their own in order to make more of a return on this investment.

And that won't just bring online more 1176s and Pultecs: there's no reason in principle why the system shouldn't be applied to reverb plates, effects units and even miked-up guitar amps, as well as rare custom outboard that simply isn't available to buy. And one of the reasons why we see quite a bit of modern high-end outboard in the launch rack is that it's an effective showcase for makers. If you're a boutique manufacturer who produces a few tens of units a year, the advantages of making your equipment available to try online are obvious.

A taste of the treats to come can be found in the new Nashville server, which is about to come online as I write, and which introduces a spectacular selection of mainly vintage gear from the collection of Grammy-winning engineer, mixer and producer F. Reid Shippen. As well as pairs of Chandler LTD-2 and Altec RS124

compressors, the highlights include pairs of super-rare Neve 1057, 1058, 2057 and 1064 equalisers and a pair of Pye compressor/limiters. Reid's website at www.robotlemon.com is well worth a look, if only to gawp at the astonishing full list of kit in his studio!

In Use

Some products are a lot simpler to use than they sound, and Access Analog is a prime example. The Analog Matrix plug-in window might look intimidating at first glance, but it really isn't. If you're accustomed to working with plug-ins, you'll be up and running in no time, and the fact that your audio is being sent halfway round the world to be processed is quickly forgotten. Apart from the lag, it really is just like working with software-controlled hardware in a local rack. Analog Matrix also adds parallel processing capability

» to everything, which opens up some interesting possibilities.

If you're coming from an in-the-box background, you might also be surprised by how some of this equipment sounds. The LA-2A compressor is a prime example: there are plug-in emulations that I like, including UA's own, but none of them sounds quite as righteous on a bass guitar as the real thing in AA's rack. I've not come across any plug-in versions of the early Neve equipment in the Nashville server, all of which has a very distinctive and likeable sound character, while the Pye limiter will bring a smile to the face of even the dourest mix engineer. I thoroughly enjoyed using some of the other equipment in the racks, too, especially the Manley and Highland Dynamics compressors. It's fair to say that a few days spent using Access Analog has lengthened my personal wishlist to an alarming degree!

However, it also reminded me why buying esoteric outboard is not for the faint-hearted. There were clearly still one or two technical gremlins in the Nashville Neves, which I hope will be ironed out by the time you read this; and with some of the vintage and valve gear, you can't always rely on perfect gain matching when two mono units are used on a stereo track. It might be nice if the trim fader could be adjusted separately for left and right channels. I also experienced one or two crashes relating to the plug-in, including a repeatable one when recalling presets for linked mono processors. However, I'd emphasise that none of this really had much impact on the real-world usability of the system, and it never wasted a significant amount of AA credit.

How much will it cost you to do a mix with Access Analog? That's quite a difficult question to answer, because it will depend a lot on how you like to work. If you load some gear into a rack, forget

about it and go off and do something else, you can easily burn up credits to no purpose. Equally, if you are focused and disciplined, it's possible to get a surprising amount done with relatively few credits. My broadband wasn't up to using the system in real time, so I typically concentrated on the main mix sources in turn, rendering the processed audio when I was happy with my settings. Working this way I got through an average of perhaps 20 credits per hour, but this was very variable. Overall, I'd say it compares favourably from both a financial and a mental health point of view with the alternative of hiring in a load of outboard, wiring it up to your studio and not sleeping for a few days!

Access Analog's technology is extremely clever, but what's equally

■ The new Nashville server draws equipment from the selection of F. Reid Shippen, including a wealth of vintage Neve modules.

impressive is how user-friendly it is. Arguably, in fact, they've made it easier to use an 1176 on the other side of the globe than to patch one in at home. It already offers real value for almost any mix engineer, and the potential for future development is obvious. Much more than just a way of introducing some analogue class into your mixes, it's a shop front for boutique manufacturers, a way to audition new gear and a playground in which you can have endless fun. If AA ever decide to add an extra A, it should probably stand for 'addictive'... ◢◢◢

$ From $15
W www.accessanalog.com

Korg Volca Sample 2

Digital Sample Sequencer

Korg rework their popular Volca Sample with some welcome additions.

ROBIN VINCENT

In his review of the original Korg Volca Sample in February 2015's edition of *Sound On Sound*, Paul Nagle believed it to be "the most versatile Volca yet," and he's not wrong. With the aptly named successor, the Volca Sample 2, it retains everything Paul liked about it and all the things he didn't because it is almost entirely the same machine. The only noticeable changes to the exterior are a USB port that has been added in such a way as to make you believe it was always there, and the case is now a more subdued and groovy dark grey as opposed to the somewhat medicinal white of the former. All the other changes are under the hood, so let's pop it open and have a look.

Recap

If you're not familiar with the Volca Sample then let me tell you that it's a fun little digital sample-playing beat machine. It has a 10-track sequencer of 16-steps and a 'Motion Sequencer' where individual parameters like sample length, speed, level, pan and envelope can be recorded for each step of each track or smoothly automated across the pattern. It has an 'Analogue Isolator' built in, which is essentially a low- and high-pass filter masquerading as Bass and Treble controls plus a rather funky reverb. It's fast, fruity, full of life and character. It doesn't do any sampling itself but with the new USB port, transferring samples over should be a piece of cake.

USB Port

The biggest new feature is all about the USB port. No longer do you have to patch the audio output of your computer to the Sync In of the Volca Sample to attempt an often hit-and-miss transfer of sample banks as data across an audio cable. With the dedicated Volca Sample Librarian software "sending your samples and patterns back and forth is now an instantaneous and pain-free process". Sadly, it wasn't for me. When attempting to pull over the samples and patterns from the Volca it complains about a lack of Korg USB Driver and then times-out during transfer. Once I'd found the driver on the website, installed, restarted and retried, the Volca Sample 2 had disappeared altogether. With the help of Google and a video by Tim Shoebridge it all worked out in the end, and I can now experience the "instantaneous and pain-free process" as my Mac OS-based colleagues are undoubtedly already doing.

Librarian

The USB port enables the use of the Volca Sample Librarian software on a suitable computer. It sucks in all the patterns and samples from the Volca and displays them in a list where you can rearrange them, audition samples and load or save new ones.

On the Patterns side you can save, load and rearrange the order of the 16 internal patterns. It comes with 12 quite stunning ones and so you might want to save them just in case you mess them up. You do have four blank patterns to play with, though.

On the Samples side it comes with 150 brand-new samples. It's a quite brilliant selection of energy-infused urban and lo-fi hip-hop sounds which, along with the patterns, gives you the sense that you are

never going to come up with anything as good or exciting as this. But in case you want to try there are 50 slots free to fill with your own library before overwriting the factory ones.

Importing a sample is simply a matter of selecting a slot and importing from the menu. The Librarian will downsample any audio file to 31.25kHz to give it that Volca Sample 2 crunchiness. You can keep adding samples until the memory is full or you reach 200. The factory bank takes up about 81 percent of the memory and most of them are under a second long. The memory, by the way is double that of the original at 8MB, which amounts to 130 seconds of sample time.

Once you are happy with the order of things you can punt it all across to the machine for playing. It takes moments to send over a sample or two and about a minute for a full dump depending on the size of your samples. I can certainly confirm that the process is pain-free and often instantaneous.

Chain Function

The most significant workflow development comes in the form of pattern chaining. Instead of loading a single pattern sequence to play with, you can now chain up as many as you want to play one after the other which potentially transforms Sample 2 into a 256-note sequencer. Although isn't that what Song Mode was on the original Volca Sample? The sequence steps are labelled M1-M16, which refers to the 16 memory slots for patterns. On the original they were labelled M1-M10 for the 10 memory slots and then S1-S6, which were for saving songs. Songs

The Librarian software makes importing and managing samples a lot easier.

were a collection of patterns that you had arranged to play back in order and Chain Function seems to have replaced it.

I can't quite decide whether the removal of Song Mode in favour of extending the sequence count from 10 to 16 is a good or a bad thing. The versatility of Song Mode was that you could arrange the 10 sequences quickly and easily on the device itself. The Chain Function benefits from access to 16 sequences, but the order can only be changed in the Librarian. Also, with Song Mode you can save six songs with their arrangement of patterns whereas you don't save the Chain Function, you just turn it on and select a series of sequences.

It would be nice to have both 16 sequences and a Song Mode but Korg must have had to make a judgement on what customers were asking for and what could be achieved in the context of the little box.

Step Jump

Korg has added another option to Step Jump Mode. This is a finger-tastic mode where the sequence will jump to whatever step you tap and stay there on repeat if you hold your finger down. You can do this with multiple fingers and it will jump around between them in a thoroughly entertaining way. The original mode had the playback move on from the last released step whereas now you can opt for the steps to continue and pick up in the appropriate place when you release your finger. This brings a lot more flow to the mode and makes it feel more like remixing within

the context of the pattern rather than as a disruption to it. The only snag is that it's a global mode choice and so requires you to turn the box off and on again to swap between them.

Start Delay Function

The Volca Sample 2 has a decent Swing engine for dialling in some feel and groove to the playback. However, with the Start Delay function you can craft in a bit more nuance. Start Delay simply delays the start of the sample by a small amount from 0-66, which appears to be an arbitrary fraction of a step.

You can apply this along with any of the Step Programmable Parameters to any step of any part by being in Step Mode with the Record button on and holding the sequence step while turning the appropriate knobs.

Conclusion

Version 2 of the Volca Sample builds on what was already a killer little box. Without question the USB port and Librarian makes the samples and patterns easier to manage and can turn it into a sound source that's uniquely you. New users are not going to miss the Song Mode and we all benefit from the increased memory and 16 sequences. It's versatile, fun, and the quality of the new factory bank is superb — it's a cracking little sample production machine. ▰▰▰

$ $149.99
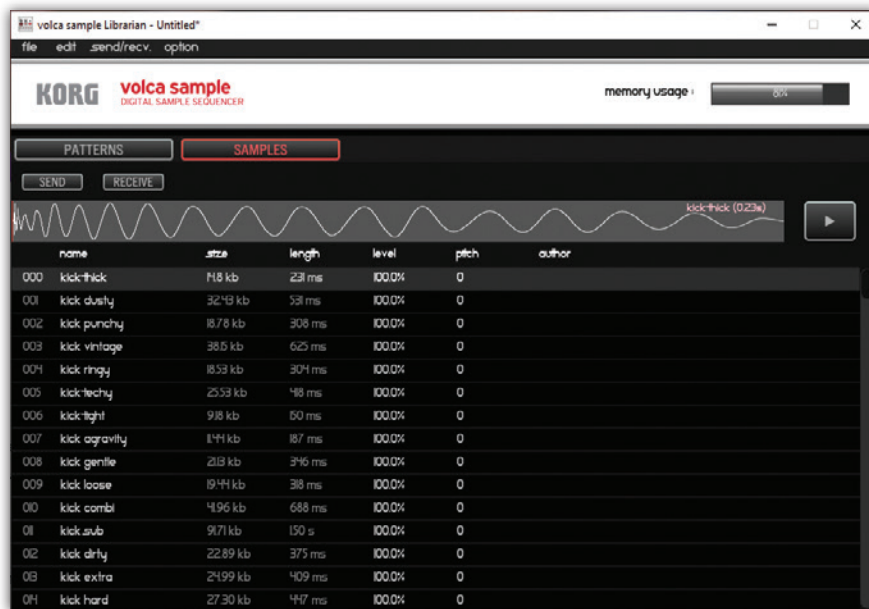W www.korg.com

---

Korg Volca Sample 2
$149

PROS
• Two hundred sample slots.
• Sixteen patterns.
• Increased memory.
• Easy sample management.
• Direct MIDI control over USB.
• Enormously fun.

CONS
• Doesn't sample.
• Volcas are a bit small and fiddly.
• No Song Mode.

SUMMARY
One of the best Volcas just got more memory, a fat new factory bank and a Librarian for easy sample management. It's an awesome pocket-sized sample production station.

Zoom
LiveTrak L-8

Mixer, Multitrack Recorder & USB Audio Interface

Could this be the perfect portable package for podcasters and musicians alike?

MATT HOUGHTON

Zoom's LiveTrak L-8 (I'll call it the L-8 for short) is aimed at a very broad audience with a range of different workflows but, while an affordable hybrid product like this will inevitably lack some features for some users, I reckon Zoom have managed to strike a great balance between sonic quality,

functionality, ease-of-use and price. So let's take a closer look at what this thing can do — and what it cannot — starting with its role as a digital mixer, before moving on to the USB interfacing and standalone recording side of things.

Ins, Outs & Mixing

As a digital mixer, the LiveTrak L-8 is fairly simple, in that all the input channels are

permanently routed to the master stereo bus. There are no subgroup busses or, other than one that goes directly to the internal effects engine, aux sends. That effects send is available to all channels except, obviously,

pad. 48V phantom power is switchable globally for the six mic inputs; there's no LED indicator but a message flashes on the screen when phantom is engaged, or if it is engaged when you switch the unit on. Finally, channels 7 and 8 cater for a number of different sources: each can receive mono line-level sources (TS jack), a stereo USB out from a connected computer, or samples triggered by the six Sound Pads (see box). These two channels can also, when used in combination, receive a stereo analogue line signal on a TRRS mini-jack, with a 'mix minus' monitoring function to prevent feedback, providing a way to connect smartphones for podcasts, video conferencing and so on.

The main stereo mix comes out on two balanced XLRs, and that signal is duplicated on the first headphone output. It's also available on three more headphone outputs, each of which, like the first, has a dedicated level control. These three can be switched to monitor one of three secondary mixes, Mix A, Mix B and Mix C, of which more below. The headphone amps sound clean and clear and do the job well for most headphones, though with high-impedance models I found there wasn't lots of gain in reserve — occasionally frustrating when I was miking quiet sources, but it won't be an issue for most users.

Each of the eight input channels has its own fader, mute (but not solo) and Select buttons, and when a channel is selected you can tweak further parameters in the Channel Strip section. This has five continuous rotary encoders and a (12dB/octave 75Hz) high-pass filter button. Each encoder is circled by 13 LEDs, but as each LED has five levels of brightness

Zoom LiveTrak L-8
$399

PROS.
• Lightweight and will run off batteries, USB bus power or AC supply.
• Sounds clean and pretty quiet, even at high gains.
• Useful reverb options.
• Four headphone outputs.
• The price is right.

CONS
• Can't access USB, phone and Sound Pads simultaneously.
• Cannot EQ the main mix.
• Support for 96kHz is limited.

SUMMARY
The LiveTrak L-8 strikes a great balance between quality, versatility and ease of use.

Effects

There are six effect types available, though you can have only one active at once. The top four function buttons give you Hall 1, Hall 2, Room and Plate reverbs. The bottom row offers Delay, Chorus, Vocal 1 and Vocal 2, the latter pair being a combination of reverb and delay. Pressing the relevant button allows you to edit two parameters for the effect: Tone and Decay for the reverbs, Time and Feedback for the delay, Tone and Rate for the Chorus, and Time and Decay for the two Vocal effects. It's not a lot of control, admittedly, but between the channel send level, these parameters and the EFX RTN channel fader, I found it sufficient.

---

and they light in combinations you're treated to a finer level of indication than first sight suggests. A very minor gripe is that the height of the knobs means they obscure some LEDs' default positions (eg. centre pan) when you're seated, with the L-8 on your desktop. The encoders control panning, the effects send level and a three-band EQ, comprising high (10kHz) and low (100Hz) shelves and a peaking mid band (2.5kHz). Each band can boost up to ±15dB, again in fine enough steps that you have plenty of control.

Despite this being a digital desk (meaning no physical routing limitations in between the converters) you can't apply EQ to the effects return or master mix; I can't say I missed that ability but some might. Naturally, you can't apply reverb to the main mix, since there's only one effects engine and its signal already flows to the mix bus.

A feature I wish had been included but is lacking is a polarity inverter for at least one or two channels. A compressor or two would also have been welcome at times when using this as a standalone device. Perhaps such things could be implemented in a firmware revision, by giving some hardware controls a secondary function?

Making A Scene

As mentioned in passing, you can create up to four different post-fader mix balances — the main mix, Mix A, Mix B and Mix C — and each can be sent to its own headphone output if desired. So it's possible to use these to create artist monitor mixes or to feed external processors/effects (you'd have to return them on spare mixer channels). You can copy and paste from one mix to another, making it quick and easy to set up personalised cue mixes. The faders aren't motorised, but switch between mixes »

the effects' stereo return channel and the stereo master mix channel. However, some behind-the-scenes digital sophistication means that it is still perfectly possible to create and distribute different cue mixes, or send signals to external processors/effects.

In addition to the master stereo channel and the stereo EFX RTN, the mixer has eight input channels. The first two accept mic, line or instrument sources, the last selected using a Hi-Z button above the gain pot. The next four can be used with mic or line sources and feature a -26dB

The SDHC/SDXC card for the multitrack recorder is inserted in a compartment at the rear, where you'll also find the on-off switch.

and the level meter LEDs will helpfully light to indicate the 'virtual fader position' for each channel, and turn off when the fader is at the correct position. As this device costs less than many eight-fader control surfaces, I reckon that's a the perfect balance of cost and convenience.

It's important to understand that the Mixes store only the fader settings: the Gain, Pad/Hi-Z, Select, Channel Strip and Mute status aren't recalled. But more settings can be stored in what Zoom call Scenes. Several settings are stored in each Scene: the fader positions (for every channel, including EFX RTN and Master); Mute status; all Channel

Sound Pads

The six Sound Pads are essentially sample triggers, three playing back on channel 7 and three on channel 8. The default sounds are the sort of effect typically used in radio shows and podcasts (applause, laughter, cash register, drum roll, sad trombone, scratched record) but there are others to choose from (in Mixer mode press the Sound Pad function key) and you can import or record your own samples too; it's a feature that will appeal to many podcasters. Note, though, that when monitoring other sources (eg. USB or a TRRS phone source) on channels 7 or 8, you can access either no Pads or only half of them, and you can't use the Sound Pads in 96kHz mode either.

Strip settings; the selected effect patch and parameters; and the input source selection for channels 7+8.

Hit the big Scene button at the bottom right, and the eight multi-function buttons beneath the screen give you direct access to seven separate scenes which can be stored and recalled. Press a button and you have the option to save/overwrite or recall the Scene or cancel to take you out of the menu. The eighth button allows you to reset the selected Scene. It's a very helpful facility and really intuitive. It might have been nice to be able to recall the channel Select status too, and the Rec/Play status for the multitrack recorder (discussed below), but those aren't mission-critical omissions.

USB

Many potential purchasers will be primarily interested in the L-8's recording capabilities. It can be used either as a USB 2.0 audio interface or as a standalone multitracker, recording directly to class-10 or higher SDHC (up to 32GB) or SDXC (up top 512GB) cards. You can record at 24-bit 44.1, 48 or 96 kHz to a card, though 96kHz isn't supported over USB. Neither can you use the Sound Pads, send effects, EQ or the multitracker's Overdub mode, at 96kHz. There's a class-compliant USB driver for iOS, and for Mac/Windows you download a driver from Zoom's website. Once installed, ensure the

L-8 is in Mixer mode, then press function button 8, labelled Audio I/F, and you're good to go.

Your computer is presented with 12 inputs. The eight input channels' signals (7 and 8 each have left and right channels) are taken directly after the preamp, before the Channel Strip and fader, while the stereo mix is captured post-fader. So you can record the clean input signals and tweak settings on playback, but can also capture the live stereo mix, with all your EQ, effects and fader moves applied. This way, it makes good sense for recording projects, for live streaming, and recording live performances while also delivering monitor and PA mixes. The L-8 can receive two stereo feeds from the computer, one on channel 7 and the other on channel 8. Note that selecting the USB return on either channel will limit your access to other sources (eg. Sound Pads) that are exclusive to those channels.

On The Record

While USB recording is handy, the standalone recorder side of things really won me over. It's just so quick and convenient — a joy to use. It's the default mode at startup, but you can access it at any time by pressing the big Recorder button. In this mode the function buttons serve as transport and tempo controls, and access a Setting menu.
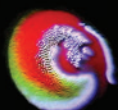
Initial recording tests are intuitive. You record-arm the desired tracks using their Rec/Play button (which cycles through three states: record-arm, multitrack playback, and off, whereby the input source is routed to the mix bus during recording/playback. Hit the Record button followed by the Play button: a new project will be created automatically as recording commences. Change the track state to Playback and hit the Rwd button, and you can audition what you recorded and adjust the channel settings to taste.

You can also navigate the project timeline by turning the Select Wheel (a rotary encoder with integral push button) and press it offline or during playback/record to insert up to 99 markers per project; the Fwd and Rwd buttons take you to the next marker in either direction. It's a great way to mark unhelpful noises or fluffed lines that need attention. It's worth noting, though, that pressing the button produces a small acoustic click, so if using this while recording you'll want to keep your mics at a safe distance.

Hitting Setting (function 4) accesses the menu. It's easy to follow but anything you don't understand will be clearly explained in the manual. Here, you can create, delete and rename projects, assign recorded or imported sounds to different tracks, scroll through a list of markers and jump to any one. You can also specify the recording format, and engage auto (threshold-triggered) record and/or 2-second pre-record modes. And you can set different playback modes, so playback stops at the end of a project or at the end of all projects, or loops around one or all.

The metronome is a welcome feature for those laying down music projects one track at a time. You're offered control over the sound (Bell, Click, Stick, Cowbell or Hi-Q), the pattern and most basic time signatures. The time-signature options are adequate but not extensive; there's no 9/8, for example. A Tempo button provides a tap-tempo facility and allows you to use the Selector Wheel to set the tempo between 40 and 250 bpm with a resolution of 0.1bpm.

There are more menu functions than I've described here, all of which are fully detailed in the manual, but you get the idea: if your needs are conventional, Zoom have you covered; if they're more esoteric I'd suggest that you consult the manual to ensure they're fully catered for.

Another function button (8) puts the recorder into Overdub mode, which

determines if recording will be overdubbed in the current project or a fresh project will be created for the recording. Note that this mode doesn't support 96kHz recording; I don't know why that is. It does include latency compensation, though, so you can monitor your performance and the backing in perfect sync, and things will line up perfectly on playback.

Punching in/out is as easy as could be. You navigate to the section where you want the performer to start singing/playing, record-arm the tracks on which you wish to punch in, and start playback. Hit the main record button to punch in, and hit it again to punch out. You can punch in/out up to 10 times during each playback pass. Between this and the marker system, dropping in corrections becomes second nature very quickly.

Playback or recording is not possible while you're in the Setting menu, and having to exit the menus before hitting record/play sometimes made me pause for thought. Similarly, once playing back or if you have the record button engaged, ready to record, you cannot change the recording/playback status of individual tracks. When first using the L-8 I often hit Record and then realised I needed to change the status of one or two tracks. These are both very minor frustrations, though, and I'm sure with more 'miles on the clock' any sense of annoyance would evaporate.

As with USB recording, each of the eight input channels can be recorded pre- the Channel Strip and fader, and the post-fader master mix signal can be captured. Again, the only channel that can't be recorded is the effects return; I'm not entirely sure why that is but it's not a particularly important omission. It's also worth noting that the L-8 can record via USB to your DAW and direct to the SD card simultaneously, and that a card reader allows your computer to access recordings made to the SD card, or to deposit sounds there for import to recorder tracks, or to the Sound Pads.

Lasting Impressions

As I hinted at the outset, I love the Zoom LiveTrak L-8. It offers so much convenience and versatility for so little money, and it sounds decent too. The mic preamps, which have a gain range of +10 to +54 dB, are never going to be world-beaters on this sort of device, but they are competent: clean and quiet and as good as any in their class. I was happy using them for exposed sources like spoken word (eg. podcasting) and when applying plenty of

The USB 2.0 port, tucked beneath the L-8, has a good strain-relief system, and can be used for communication with and bus powering by a computer, or to hook up to the supplied mains power adapter or a USB battery. Alternatively, four AA batteries can power the L-8.

gain for dynamic mics (a Shure SM57 and Heil Sound PR40) there was pleasingly little hiss. The neutral-sounding broad-brush EQ is exactly what's required here, and I found a handful of the effects (mainly the reverbs) very usable, even if I always wanted to tweak the default settings to something that sounded less obvious.

Operation generally is super easy: the manual, though helpful, isn't essential to get started, and the on-screen menus are mercifully shallow. When switching between modes there are no pops, pauses or glitches. There are enough headphone outs for serious projects. My list of gripes is very short, the most notable being the lack of EQ for the main mix, and the source-routing limitations for channels 7+8. And they're more than offset by the L-8's convenience, versatility and portability.

It's really very hard to make something as complicated as this seem so effortless. And better still, it's a lightweight (a shade over 1.5kg) portable package with a small footprint, yet gives spacious access to controls. The fact that it can run off batteries, USB power blocks, or bus power just makes it better still. So I take my hat off to the designers, and will file the LiveTrak L-8 under 'bloody impressive for the price'. ▰▰▰

$ $399.
W https://zoomcorp.com

Only pros.
No cons.

Nugen Audio Paragon

Convolution Reverb

Have you ever wished convolution reverbs could be a little more malleable?

HUGH ROBJOHNS

I f the name Paragon sounds familiar, it might be because you read Sam Inglis' article in *SOS* November 2020 (https://sosm.ag/Paragon) about the genesis of this Nugen Audio plug-in. And if you haven't read that article yet, I'd urge you to do so before reading this review, as Paragon is no 'ordinary' reverb plug-in and Sam's article provides helpful background to its development and underlying technology.

Acting On Impulse

As many other reverbs are now, this one is based on convolution. But, while practical convolution reverbs have been around for over 20 years now, Paragon is no ordinary convolution reverb either. Based on groundbreaking research and three years of intensive development by Dr Jez Wells of the University of York and Nugen's own software geniuses, Paragon employs additive resynthesized impulse responses. This is a highly innovative technology which Sam explains well in his article, but the brief version is that impulse responses captured in real spaces are analysed to create a model which can be modified through a process of resynthesis.

This intriguing technology enables complex reverberation fields to be generated which combine the exceptional sonic accuracy and realism of convolution reverbs, but with the ability to be tweaked and modified in the same way as traditional algorithmic reverbs. So fundamental parameters that shape the character of a reverb, such as decay time, room size, and HF and LF damping, can all be adjusted on the fly, almost in real time. Because the reverberation field is produced by individually resynthesized impulse responses, changing parameters doesn't result in the processing artefacts that blight systems that modify impulse responses through, for example,

time-stretching techniques.

You'll probably have noticed that I said 'almost' real time. Parameter adjustments do inherently require a brand-new impulse response to be calculated (the resynthesis part) before it can be used in the convolution process, and that recalculation obviously takes a finite time. Nugen's Paragon plug-in very cleverly subdivides this resynthesizing workload into separate calculations for different frequency bands, and for early reflections separately from decay tails, too. All of this dramatically simplifies and speeds up the resynthesis of new impulse responses.

When a reverb parameter is changed, the Nugen badge at the top right of the plug-in window spins around while the number-crunching is being done, but even on my ancient Windows 7/64 computer it's usually a case of 'blink and you miss it': typically it only takes a few tens of milliseconds to recalculate and implement the IR updates, and in practice I found

recalculation delays to be insignificant, and certainly no impediment while fine-tuning a reverb.

Making Spaces

Like any convolution-based reverb, the key to Paragon's sound quality lies in the underlying impulse responses. It currently ships with 49 unique IRs, captured in a wide variety of acoustic spaces, and the Nugen team have analysed these source IRs to create the models which allow the IRs to be customised through additive resynthesis to suit any desired requirements. Of course, this analysis stage of the technology means, sadly, that Paragon can't run any old IR — you can't import your own.

These factory preset IRs cover the usual bases of halls, chambers, churches, rooms and so on, and there are bright, dark, large, medium, and small alternatives, too. Slightly oddly, there is no Plate reverb IR here, but there is an interesting range of unusual acoustic spaces including car and van

Both the amplitude (left) and decay (right) can be adjusted differently at different frequencies.

interiors, a library, a hotel corridor, the room next door (very handy for anyone involved in radio drama production!), stone and tiled rooms, a stairwell, a telephone and many more besides. The first three preset options are also unusual in that they 'Add Presence, Warmth or Width'. See the box for a complete listing.

Apparently, Paragon's IRs were all captured as Ambisonic recordings because the B-format signals can be processed to generate any number of output channels, arranged in any layout. Consequently, Paragon can accommodate anything from single mono to 7.1.2 channels (the latter suiting Dolby Atmos sound beds, for example), all from the same core IRs.

Another unusual aspect of Paragon is a parameter called Crosstalk. With the Crosstalk control dialled down to zero, the convolution reverb engine is essentially multi-mono, meaning that an input signal only generates reverb for its corresponding output. So a left channel input only creates reverb on the left channel output. That isn't exactly the normal arrangement (although it could be a useful effect in some situations), but as the Crosstalk control is increased each input signal is allowed to bleed into its adjacent channel(s) to generate reverberation from those outputs too. So in a stereo configuration with a sensible crosstalk level, the left input generates reverb spreading across both the left and right outputs in the usual way. The precise amount of crosstalk being sent from, and arriving in, each channel is

shown as a percentage in the GUI, and it is an interesting and unusual facility with which to experiment.

The Main Page

The plug-in's GUI has four pages, but the most used controls are found on the main page. A second page shows the specific impulse response characteristics and offers controls to change the frequency response and frequency-dependent decay rates. The third page deals with the I/O configuration, and a fourth covers the various user-setup options. The idea is to give you plenty of control, but that once you've configured the plug-in appropriately for a project, most of your reverb selection and tweaking can be done directly from the main display page. It works well.

Despite the sophistication and versatility of Paragon's IR engine, this main screen feels pretty familiar and intuitive. The window is dominated by a large circular 'Reverb Surround Scope' with adjacent 'node' buttons acting as either Solo or Mute buttons for each output channel. The Scope display illustrates how the reverberation spreads to the output channels, with its brightness indicating volume and colour differentiating the wet and dry elements. It's an unusual way to view reverberation, but I can see why Nugen adopted it: with experience it becomes rather informative.

Pairs of vertical sliders either side of the Scope adjust the core parameters of Pre-delay and Decay times, room Size and Brightness. The Decay and Size parameters can be scaled up to

five times the IR's original values, and all parameter values can be typed in, if preferred. Across the top of the window a horizontal slider sets high- and low-pass filter frequencies while another alters the amount of inter-channel Crosstalk, with buttons to select the relevant channels and directions, when appropriate.

Below the Scope section, and present across all menu screens, another pair of sliders adjust the wet/dry mix and the output Trim level, with bar-graph meters for each channel's output level. At the very top of the GUI are all the usual Bypass, undo/redo, and A/B memory comparison buttons, along with a preset selection

»

Nugen Audio Paragon
$599

PROS
• Innovative technology takes convolution reverb to a new level.
• Very varied collection of superb-sounding IR acoustic environments.
• Support for a vast array of multichannel formats.
• Largely familiar controls for parameter adjustments.
• Comprehensive global and channel-based reverb parameters

CONS
• Some minor grumbles about the GUI.
• Can't use with personal IR files.

SUMMARY
Paragon is the result of advanced technology combining additive re-synthesis with convolution reverb to bestow the sound quality and accuracy of a multichannel convolution reverb with the custom tweakability of an algorithmic reverb.

dropdown. The Nugen logo here converts to a spinning wheel when IR recalculations are being performed.

Diving Deeper

For most, the second GUI page will feel the least familiar. It is dominated by two views of the selected impulse response: the upper one displays the IR as a spectrogram, plotting frequency versus time, while the lower one is a conventional frequency response showing frequency versus amplitude. These are used to illustrate the selected IR's characteristics and any changes to its frequency response and/or frequency-dependent decay times.

As the parameter controls are shared, buttons to the right of the plots select whether they are applied to the amplitude or decay time, while more buttons below select which channel(s) are affected. Parameter adjustments are made either by manipulating the EQ curve shown in the lower chart, much like most EQ plug-ins, or by entering numerical values in the relevant parameter display boxes below. There are options here to change the shape of the EQ curve (bell, or high/low shelves), frequency, Q (bandwidth), and value, this last being either the gain amount for frequency response changes or a time multiplier for decay-time adjustments. Up to four EQ bands can be implemented either by double-clicking on the EQ curve or by using the add/remove/select buttons at the bottom left of the lower plot.

A Complexity slider effectively adjusts the density of the resynthesized IR, with higher values noticeably increasing recalculation times. And a Reset button removes all current user adjustments to restore the original IR preset.

Usefully, five 'test sounds' are also included on this IR page as an aid when fine-tuning reverb settings. The options are acoustic guitar, kick drum, harmonica and both short and long vocals. When selected, these loopable samples replace any audio playing from the DAW.

The I/O page displays various parameters for the individual audio channels in the current configuration, and these parameter adjustments act in addition to those dialled up on the main page. The channels are arrayed horizontally, starting with a Solo/mute button and a coloured bar-graph meter (with a peak-level readout) which is switchable between input and output levels. White arrows above/below each

Factory Presets

1-3	Add Presence/Warmth/Width	29-30	Office 1/2
4	Air Room	31	Public Library
5-7	Bright Chamber/Church/Room	32-33	Room Ambience/Next door
8	Car Interior	34-35	Slapback Long/Short
9	Clear Vocal Hall	36-43	Small Chamber/Church/Club/Room 1/
10 -13	Dark Chamber/Church/Hall/Room		Room 2/Stone Room/Theatre/Tiled Room
14-18	Hotel Corridor 1/Corridor 2/Lobby/	44	Stairwell
	Room 1/Room 2	45	Sustain Hall
19-23	Large Bright Hall/Chamber/Church/	46	Telephone
	Club/Theatre	47	Van interior
24-28	Medium Bright Hall/Chamber/Church/	48	Warm Chamber
	Stone Room/Tiled Room	49	Zeppelin Drums

As well as covering most expected environments, the user library includes a number of quirkier settings that should prove useful in radio drama and sound for picture, such as the 'Room Next Door' preset pictured here.

meter adjust the input and output levels, although the attenuation/gain setting is shown right at the bottom of the window; something I found frustrating as it's nowhere near the pointer being moved! There's also no capability of typing in a specific gain value. Other per-channel parameters comprise a slider to adjust the high- and low-pass filters, and numerical boxes for pre-delay, decay-time multiplier, and crosstalk levels into and from the channel. There's also a button to toggle the wet signal on and off for each channel.

Useful and powerful though these individual channel parameter adjustments are, I think there's a potential trap here: settings are not reflected or even warned about on the Main GUI page, and I caught myself out once, wondering why it sounded like there was pre-delay when the global slider was set at zero! I'm sure some means of warning that individual channel parameters differ from the global settings could be easily implemented — perhaps by altering the parameter label colours on the relevant sliders, for example.
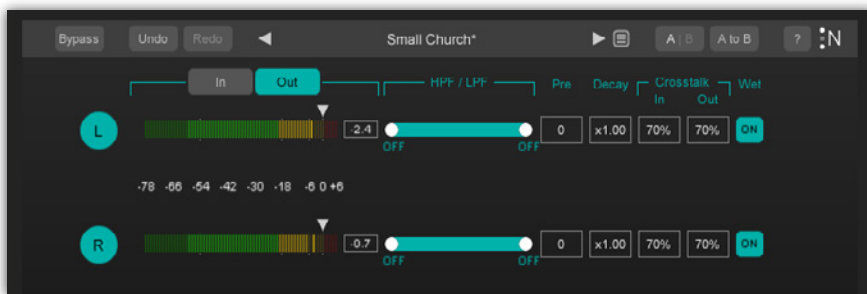
The IO window usefully offers the ability to specify crosstalk between channels.

The final GUI page, selected with the cog icon, configures input and output channel routing (defaulting to an automatic mode), determines if the node buttons act to solo or mute the respective channels, select whether pre-delay times are shown in ms or samples, alter the replay volume of the test samples, and allow the GUI colour scheme to be changed. I felt a couple of things here could be improved. First, although customisable colours are available for different meter regions, this only affects the meters on the I/O page; those at the bottom of the main page remain resolutely monochrome. Second, changes to the GUI colour schemes only remain for the current session, and must be saved with individual presets to persist.

Verdict

The final arbiter of any reverb is, of course, the quality of its reverberation and I can't overstate how impressed I am with what Paragon can do. The source IRs are well chosen and of high quality, making the acoustic environments that can be created both convincing and natural. That's true of some other convolution reverbs, but Paragon adds almost limitless customisation with just a few parameter tweaks. The quality persists across different output formats too: I experimented with 2, 4 and 5.1 configurations, all of which produced delightfully immersive acoustic environments.

Importantly, Paragon is remarkably straightforward to use, simply because its operating paradigm is so similar to a traditional algorithmic reverb. Manipulating the decay times at different frequencies is novel for a convolution reverb, but the visually interactive approach is very effective and informative. The provision of musical samples is also handy in providing a consistent reference to judge reverb parameters.

The GUI might benefit from some small improvements, described above, but this is not a major criticism, and I'm sure this side of things will evolve.

In summary, then, I'm impressed: Paragon is a genuinely innovative and excellent-sounding reverberation processor. It is obviously an excellent choice for music mixing projects, but the Add Warmth, Presence and Width options make it useful for mastering effects as well, and the impressive range of presets will also make Paragon a very powerful tool for other audio applications, such as film and TV post-production, and radio drama.

This really is a very impressive technology that takes convolution reverbs to a new level, and I heartily recommend checking out the free trial! ▮▮▮

$ $599.
T   Nugen Audio +1(0) 888 382 8188.
E   sales@nugenaudio.com
W   https://nugenaudio.com

Avantone Planar

Planar Magnetic Headphones

If you crave the refined sound of large planar magnetic cans but are put off by the price, Avantone's newest headphones could well be the answer.

SAM INGLIS

The headphone market shows no sign of ceasing to boom. It's still dominated by traditional moving-coil designs, but alternative technologies are gaining traction too. Already this year I've been very impressed by HEDD Audio's HEDDphones, the first headphones to use an Air Motion Transformer as a driver. Elsewhere, breakthroughs in magnet technology have brought about renewed interest in planar magnetic headphones, thanks mainly to the efforts of Dr Drag Colich at Audeze. For a full explanation of how a planar magnetic driver works, go to www.soundonsound.com/reviews/audeze-lcd-x-el8 where you'll find our detailed review of Audeze's flagship LCD-X.

The latest manufacturers to embrace this technology are Avantone, US-based makers of mics, loudspeakers and other studio equipment. One of the more interesting products in their existing catalogue are the MP1 'Mixphones'; these are closed-back phones that use conventional moving-coil dynamic drivers, but they incorporate a mono switch and a voicing option that cuts out the extreme high and low frequencies so you can pretend you're mixing on an Auratone. The Mixphones have now been joined in Avantone's line-up by a second pair of headphones which, as the name suggests, use planar magnetic drivers

rather than moving-coil transducers. Like most planar designs, the Planars are open-backed headphones designed primarily for mixing and hi-fi listening.

Planning A Head

Available in red or black, the Planars are shipped in an imposing cardboard box. The included "eco-friendly carrying bag with cable pouch" proves to be a simple canvas bag that offers relatively little protection, and is no substitute for a rigid case. On the plus side, though, you get two cables, both screened with an attractive and robust braid. The shorter of the two has a mini-jack at either end and can be plugged into either of the two earcups; the longer is a Y-cable that attaches to both earcups, and always delivers the correct channel to each side regardless of which way around you connect it.

Planar magnetic phones are usually on the large side, and none more so than the Planars. The rectangular earcups are about 11 x 7 cm in diameter and fully 5cm deep with their removable earpads, while the total mass of the headphones is almost half a kilo. Despite this, they are perfectly comfortable for extended use. The metal band that bridges the two earcups is augmented by a soft, broad headband, and it's this that rests on top of your head, spreading the load. The earcups are hinged and rotate freely in two axes; however, there's no way to adjust their height relative to the headband, and they sat a touch lower on my ears than I'd have wanted.

Strong Points

Historically, one of the drawbacks of planar magnetic headphones has been low sensitivity. Thanks to the use of rare earth magnets and improvements in fluxor design, that's no longer an issue, and the Planars offer a more-than-respectable sensitivity of 104dB/mW. They present a nominal impedance of 32Ω, and like most planar designs, claim a very low distortion figure, with THD measuring less than 0.1 percent at 100dB.

From a sonic point of view, planar designs also have historic strengths and weaknesses. They can put across low frequencies in a way that most moving-coil phones struggle to match, and the presentation of musical dynamics in a good pair of planar phones is hard to beat; but achieving a high-frequency response that's flat to 20kHz is

a challenge. Avantone boast a frequency response of 30Hz-30kHz for the Planars, but in the absence of any tolerance, this isn't really a meaningful claim.

When I reviewed Avantone's Mixphones, back in 2016, they seemed unusually sensitive to differences between headphone amps and, in particular, could not reproduce low bass properly when driven from a low-power source. In theory, planar magnetic headphones ought to be easy for any amp to drive, but in practice I experienced much the same effect with the Planars. Connected to the built-in headphone amp on a laptop or phone, they sounded brash in the upper midrange, whilst missing an octave or two in the low frequencies. Even the headphone amp on a cheap mains-powered audio interface didn't do them justice, and it was only when I plumbed them into my DACS Headlite that I heard what they are really capable of.

Making Planars For Nigel

What they are capable of, it turns out, is quite impressive, and although there is a sonic differential between the Planars and flagship Audeze models, it's a lot narrower than the difference in their respective costs. The same basic strengths are certainly apparent: they produce a clear, natural, open sound that, like all good monitor systems, makes you feel you're listening to the music rather than the headphones. In a direct comparison with Audeze's LCD-MX4s, the stereo image on the Planars perhaps even felt a touch wider, and although they don't present drums with quite the same thrilling punch as the LCDs, they still outperform most moving-coil headphones in this regard.

When connected to an amp with serious welly, the Planars have plenty of bass extension, but there is still a tendency for the very low frequencies to be overshadowed by a hump somewhere in the 100Hz region. They are also a touch less smooth in the upper midrange than really state-of-the-art models like the LCDs, occasionally coming across as slightly gritty. But they are as close to subjectively 'flat' as most headphones in their own price range, and for myself, I find a midrange emphasis much more useful for mixing than the 'scooped' tonality beloved of many headphone designers.

Price-wise, in fact, the Planars go head to head with Audeze's more affordable LCD-1 model, but there's actually less

similarity there in terms of sound or physical design. The Planars are bulky and heavy, with a classic planar sound that foregrounds the upper bass and the midrange; the LCD-1s are lightweight, small and portable, and their tonality is quite a bit more crisp. In fact, the LCD-1s sound a lot more like standard moving-coil headphones in comparison, and I personally preferred the relaxed, open feel of the Planars.

I strongly suspect this is down to a key feature that they have in common with the larger LCDs and the HEDDphones: earcups that are not only large, but also very deep. This positions the driver further away from the ear and, in my experience, delivers a more speaker-like experience. Until now, I've only encountered this style of earcup design on models costing north of $1000, so it's fantastic to see it brought to market at a much lower price. In these days when headphone mixing is becoming ever more important, many of us aspire to own high-end planar magnetic phones, but haven't been able to justify the cost. Whilst the Avantone Planars are still a significant investment, they offer a much more affordable way to experience 'that sound' — and if you try them, you'll understand why so many mixing engineers think so highly of planar magnetic headphones. ///
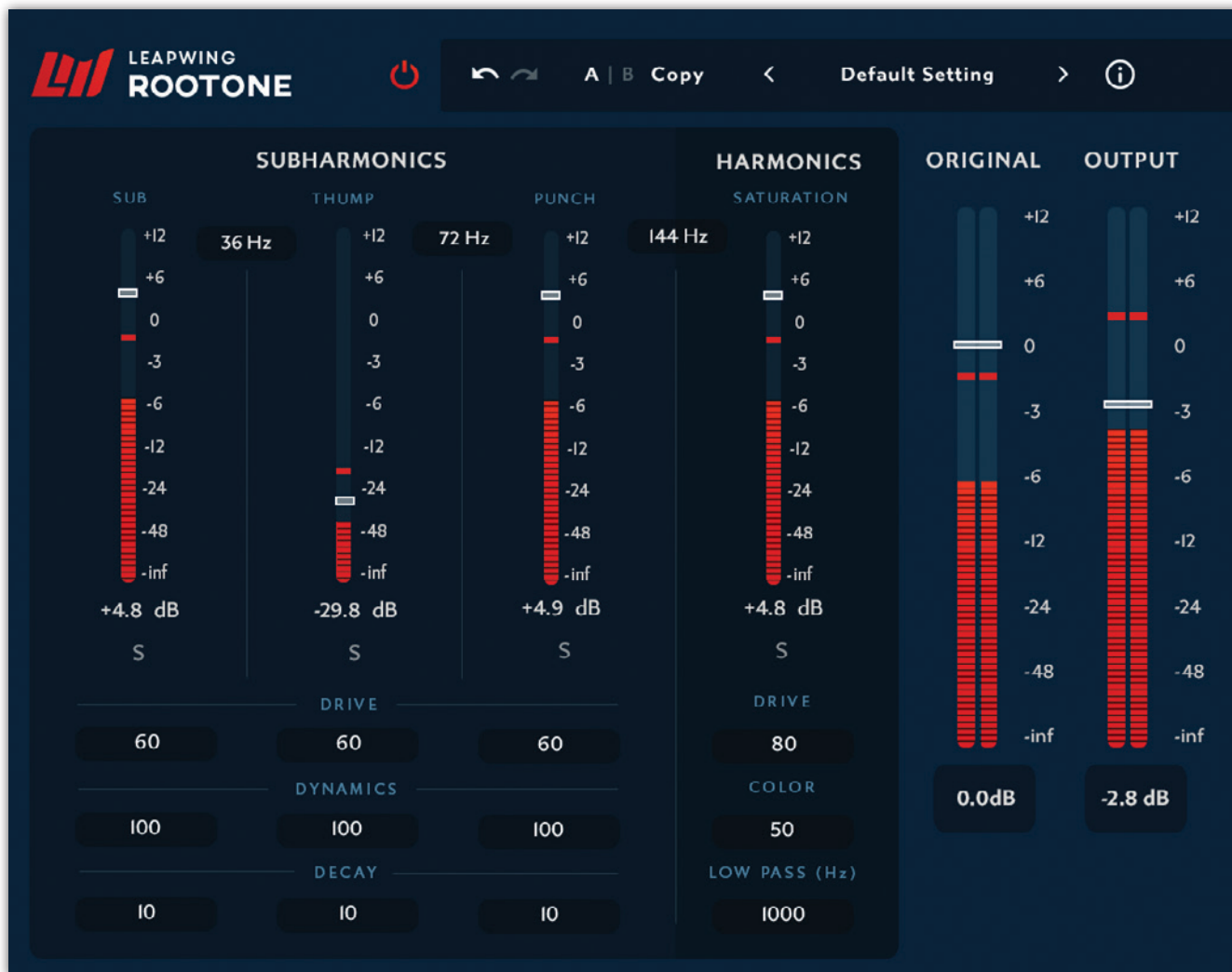
Leapwing RootOne

Bass-enhancement Plug-in

RootOne refreshes the parts most other enhancers cannot reach.

FREDERICK NORÉN

I was a little late in discovering Leapwing Audio, but when I eventually stumbled upon their DynOne plug-in a couple of years ago I loved it. I have been an avid user of their plug-ins ever since, and I particularly like that they invent new types and implementations of audio processing, rather than delivering yet more analogue emulations. So when I read about their new RootOne, a bass-enhancement plug-in

for Mac and Windows in VST, VST3, AU (Mac-only) and native AAX formats, I was excited, and couldn't wait to try it out. The world isn't exactly short of bass enhancers and sub synths (check out the Alternatives box for some examples) but, as I'll explain, RootOne does things a little differently, and the results are impressive.

Down & Dirty

RootOne's processing is divided into two sections: Subharmonics and Harmonics, and the Subharmonics are divided into three frequency bands called Sub, Thump and Punch. These three bands have adjustable crossover frequencies: Sub up to 56Hz, Thump up to 104Hz and Punch up

to 196Hz (their default positions are 34, 72 and 112 Hz, respectively). All three bands have identical controls: the fader sets the amount of added subharmonics; the Drive parameter sends the subharmonics to a separate saturation process; Dynamics controls the dynamic range of the subharmonics; and Decay can prolong their decay.

The Harmonics section is a saturation processor, which adds harmonics (based on the subharmonics) to enhance the character of the processed sound. The Drive setting adjusts the amount of generated distortion, Color sets the degree of symmetry (or asymmetry!) of the distortion, and Low Pass applies

a low-pass filter, whose turnover frequency can be set anywhere between 100Hz and 1kHz. The contribution of each frequency band can be heard in solo, as well as visually confirmed by looking at the level meters, and given how potent the process is, I strongly recommend that you do spend time listening in solo while tweaking each band.

So far, it sounds useful if perhaps also rather conventional, but there's rather more to Leapwing's process, which took them two years to develop. RootOne continuously analyses the source material to determine the frequency, phase and level of the root tones (fundamental bass notes). The Sub, Thump and Punch frequency bands react to these root tones roughly an octave above and, together, the bands generate a couple of subharmonics that are perfectly in phase with the root tone.

Where a source lacks low end, instead of adding subharmonics it's sometimes useful to add 'body' higher up in the bass region. RootOne's saturation process is great for that: it adds harmonics based on the new subharmonics, and the cool thing is that you don't have to add the subharmonics themselves; you just set the crossover frequencies accordingly.

Performance

Running some test tones through RootOne made it easy to see how RootOne adds both subharmonics and harmonics. Tweaking the individual Drive parameters along with the Drive and Color parameters

Leapwing RootOne
$199

PROS
• Easy to use.
• Intelligent sub-harmonics generation.
• In phase with the source audio.
• Can 'inflate' the sound.

CONS
• So seductive that's it's easy to overuse it!

SUMMARY
Combining phase-aligned subharmonics and harmonics in a very musical and intelligent way, Leapwing RootOne sets new standards in bass enhancement. No matter how thin-sounding the source is, the chances are that this plug-in will make something interesting out of it.

of the Harmonics section also made me realise that they all interact with each other — that's probably a reason why this plug-in seems to perform so well. Incidentally, while experimenting, I also tried sending pink noise into RootOne and, without playing back the noise itself, RootOne produces a low-end rumble that would be perfect as a spaceship background noise in a science-fiction movie!

Test tones can only tell you so much, though, and I'm sure you want to know how it performs on 'real-world' sources. During the review period, I used RootOne on a variety of different sound sources,

> ### "RootOne seemed able to do something interesting for pretty much any sound source and instrument that I was processing."

ranging from kick drums and bass lines, to tiger roars and a thin-sounding rock track from the early '90s. It was very effective in adding weight and punch to both electronic and acoustic kick drums, and with the latter it pretty much gave me the sensation of raising the fader of a pressure-zone mic positioned inside the drum, but without the spill sound from the snare drum. Perfect! With an electronic kick that lacks low-end, the remedy has often been to layer it with another, more sub-heavy kick. It's an intricate process that can change the overall 'sound' of the kick drum, and it has the potential to create thorny phase-cancellation problems in the low-end too. Using RootOne instead, I was able to add phase-coherent subharmonics not only to electronic kick drums but also to two club mixes, making them sound more interesting and 'alive' — a result that had me dancing around in the control room!

In fact, RootOne seemed able to do something interesting for pretty much any sound source and instrument that I was processing: bass lines, synths, snare drums, percussion, voice-overs... The list goes on. For example, I mocked up an intro track for the local ice hockey team and it was great fun hearing the tiger roars thunder in the PA-system, thanks to their guttural low-end. Even my efforts with challenging sources met with some success: tackling a thin-sounding rock track is always tricky, especially when there is nothing more than a stereo track to work on but, while RootOne had to work hard to find the root tone of both the

bass line and the kick drum and the result wasn't perfect, it definitely sounded better than the thin-sounding original master.

So both the capability and the sound of this plug-in are good — and having used RootOne for some time now, I reckon the control set is too. The GUI is clean and easy to read, and the controls are easy to use and intuitive: lowering the Dynamics parameter makes the added subharmonics more even in level, and adding Decay makes them more prominent.

Low Ending

Processing low-end is tricky because the sensation of an increased low-end is seductive, especially in the lowest octave. The TAD speakers and Velodyne subwoofer in my control room are phase-calibrated by a Trinnov processor and have a linear frequency response down to 20Hz. This system very accurately responded to the subharmonics generated by RootOne, in an inspiring way, so I was able to tailor the bass response to taste. If listening over a less accurate system, though, or one with more limited low-end capabilities, you should make an extra effort to monitor the result of RootOne very closely, preferably on several sound systems and on headphones too. Raising the Sub fader means adding a sub synth playing bass tones in the 15 to 40 Hz area, and that might not be audible on a pair of small nearfield monitors or in a poorly treated space, and it's easy to overdo the effect.

RootOne is a new landmark when it comes to bass enhancement and sets a high bar for the competition. I highly recommend downloading the 30-day demo version of the plug-in — just be prepared to part with some money when you fall in love with it! ▰▰▰

$ $199.
W www.leapwingaudio.com

Native Instruments
Stradivari Violin

Virtual Instrument

NI's new instrument makes owning and playing a Stradivarius easier than ever before.

ROBIN BIGWOOD

Playable, expressive solo violins are one of the holy grails of the sampling world. It's a similar problem to producing a convincing solo sax or electric guitar: how the heck to cover the sheer range of articulation and tone colours a good player continually utilises, along with the subtle but crucial transitions between the notes. And then to make it all available in a DAW, most likely driven from a standard MIDI master keyboard.

Stradivari Violin is Native Instruments' response to the challenge, developed in conjunction with Hamburg-based e-instruments. They've recorded a single violin, the silvery-sounding 1727 'Vesuvius' made towards the end of Stradivari's life, which is currently in the collection of the Museo del Violino in Cremona, Italy. The capture took place in the auditorium there, apparently involving no fewer than 32 microphones, multiple virtuosi sharing the playing duties, and security guards manning the exits (I'm not kidding).

It's worth noting too that Stradivari Violin is also a component of a larger library, Cremona Quartet, that adds a Guarneri violin, an Amati viola and a Stradivari cello, all enjoying most of the same sampling features. The entire quartet library is twice the cost of the solo violin reviewed here, and there's an upgrade path for violin owners who'd like to explore some four-way action later on down the line.

Tuning Up

First, a quick rundown of the basics. Stradivari Violin runs in NI's Kontakt or the free Kontakt Player, on version 6.2.2 or higher. It's a 23.5GB download (equating to about 39GB of uncompressed sample content) that gives you two NKI patches: a 17GB multi-mic version (occupying towards 4GB of RAM, with typical disk streaming settings), and a resource-saving fixed-mix alternative using 6GB of samples and just over 1GB of RAM. NKS controller integration extends to the usual coloured keyrange prompts and various control mappings, but the library works well with standard five-octave MIDI controllers too. Either way, all the important parameters have MIDI CCs ready-mapped, with straightforward facilities to both learn new CC mappings and also to reconfigure the keys used for selecting different articulations. I found all aspects of presentation, technically speaking, to be completely intuitive and helpful:

unlike a real violin there's a very shallow learning curve.

First Position

Putting Stradivari Violin to use immediately reveals that NI's approach has not been to reinvent the wheel, but — like so many premium-quality libraries these days — just to make a very fully featured wheel.

There are no fewer than 20 articulations on offer, all covering a G2 to F6 pitch range, beginning with a broad range of long and short bow strokes that will probably see the most use, for many projects. These are supplemented with a fine pizzicato, tremolos and trills (with variable speed), combination/repetition strokes (again variable in speed, and tempo-sync'able), along with some

nice and occasionally ghostly Special effects including Col Legno, where the wood of the bow makes contact with the string, rather than the hair. The interface lets you load any combination of eight of these articulations at a time, and a Con Sordino (bridge mute) option can be enabled per-articulation rather than globally. The mute effect is applied using a DSP filter rather than by accessing alternative sample sets, and although subtle (compared to some violinists' mutes I've heard over the years, when they've remembered to bring them on stage...) it's plausible enough.

It's possible that some users may never get round to exploring all the articulations, though, because by default both NKIs load up using an adaptive articulation scheme called Virtuoso. This selects either a sustain or marcato bow stroke depending on key velocity, while a push or pull of your pitch-bend wheel immediately accesses the short, agile spiccatissimo and staccato. That's not all: legato pitch transitions change with velocity, with a more impassioned portamento swooping in (by default) when you play harder. And the volume of the whole lot of it is subject to the position of your mod wheel. An awful lot can be achieved with Virtuoso which, if you don't require any really unusual techniques, and for a lot of 'normal' musical material, can sound realistic and expressive to the point where I think the vast majority of listeners would think they were hearing a real violinist. This is quite an achievement, already. The basic tone of the Vesuvius and character of the playing is extremely likeable: it's light, bright, airy, somewhat genteel and has a nice acoustic 'bloom' to it, and plenty of naturalistic

»

Native Instruments Stradivari Violin
$199

PROS
- A super quality capture that blends accuracy with plenty of naturalistic warmth and movement, across all 20 articulations.
- Effective, musical results, requiring only a basic MIDI controller.
- Simple and straightforward user interface and control schemes.
- Ready-mixed and multi-mic implementations give options for sound and CPU.

CONS
- Limited coverage of loud dynamic levels or aggressive playing styles.
- No support for double stops.

SUMMARY
An often surprisingly convincing, extensively sampled solo violin, of world-class provenance, that gives fine results in no time at all.

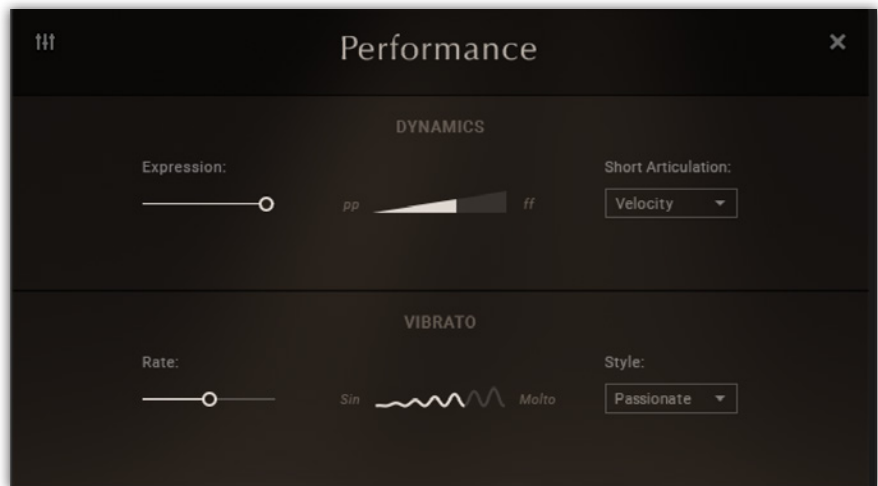movement and tone development through the sustained articulations.

Technical Studies

Digging further into the library reveals more interesting features. Aside from the eight articulation keyswitches there's one for changing bow direction for a held, sustained note. It's a nice option to have for several playing styles where simply restriking a note (especially if you do so repeatedly) can sound unnatural. Otherwise, bow direction alternates automatically for held notes when the virtual violinist runs out of bow in each direction, mostly after about four seconds. Actually, you can hear alternating strokes occurring when you restrike notes generally, with some limited use of round-robin alternation over and above this. All good of course, and crucial for authentic mimicking of a real violin. However, there's no way to override the scripting and force a particular bow direction, to allow for a characteristic repeated down-bow technique, for example, or to simply start a phrase on one direction or another.

Another keyswitch (momentary in action) forces the use of the more plaintive and vibrato-free open strings instead of fingered/stopped equivalents: a fine feature to have on hand. It's actually but one part of a sophisticated fingering position scheme with its own dedicated part of the GUI. There, and via MIDI CCs, you can force the use of warmer fingerings higher up the neck, or brighter ones low down, or simply let Kontakt figure it out in context.

Discussion of tone quality leads us nicely to the Performance pane, which handles aspects of dynamics and vibrato. As I mentioned before, the mod wheel is mapped to dynamics/volume in the Virtuoso articulation. In fact, most

non-sustaining articulations, when you select them manually, are controlled by key velocity instead. That default behaviour can be changed in the Performance pane, and

If the automatic, adaptive Virtuoso mode doesn't give you what you need then 20 separate articulations are on hand to fulfil almost all musical requirements.

you'll also find a CC11-driven Expression slider that will control overall playback level independently of any localised shaping you've recorded with the mod wheel.

Vibrato, meanwhile, is subtly sophisticated. Instead of trying to juggle lots of different sample sets with varying amounts of vibrato, Stradivari Violin applies artificial but very natural-sounding undulations to otherwise clean, non-vibrato sample sets. I say 'undulations' advisedly too, because a periodic intensifying of tone seems to happen alongside the more obvious variation of pitch. The waveform is apparently derived from analysis of real players' vibrato characteristics too, so a simple sine or triangle LFO this ain't. Both rate and amplitude are continuously variable

Dynamics and vibrato characteristics can be set up with the mouse in the Performance section, but for better momentary control they all also respond to MIDI CC messages.

On a real violin it makes a big difference to any individual note's sound quality whether it's produced by stopping low down or high up on the neck, on different strings: the Position controls let you influence that in Stradivari Violin as required.

in real time, and there are six styles on offer.

Finally, we come to the various legato options, of which there are quite a few. For the 'Long' articulations Sustain and Marcato no less than four types of legato transition are available: Fingered, Bowed, Runs and Porta[*mento*]. Fingered changes pitch without changing bow; Bowed does change bow for each new note but still sounds different to detached playing; Runs makes for faster-speaking and cleaner transitions between rapid pitch changes within a single bow; and Porta makes scooped, swoopy position changes audible between notes. It takes very little effort to explore the effects of all these different legatos, and to zone into what works best for your material. The only weakness I encountered was some short-lived phasing and doubling (which instantly shouts 'sampled!') when rapidly alternating certain pairs of notes — not much rhyme or reason which, incidentally — and when using Portamento transitions for intervals of a sixth, seventh or octave. Not all are edge-cases it must be said, but neither is any of it so serious as to mar the overall performance.

Command Performance

I'll admit a certain trepidation in taking on this review: before I'd tried Stradivari Violin my heart sank at the thought of grappling with yet another new, fiendish, fiddly articulation control scheme. It turns out I shouldn't have worried, because while there certainly are bells and whistles aplenty they're extremely easy to master and to employ to musical ends. What's more, this is a library that can give remarkably good results from the very first time of asking, that responds intuitively to playing at least as much as to 'programming', and yet will sound better still with some effort mastering its features.

My concerns are relatively few. Maximum sampled dynamic level seems to be a fairly sedate forte, not more, and a really impassioned, unbridled, hair-breaking sound is not part of its repertoire. There's no support for double-stopping: playback is strictly monophonic. And the built-in MIDI learn facilities are strictly for MIDI CCs only, so you can't easily assign vibrato amplitude (for example) to respond to aftertouch, which seems such an obvious tie-up.

In the final judgement, and assessing the extent to which the library sounds 'real' and convincing, a little perspective is needed. Attempts to pull off the solo part of the Brahms concerto will probably get sniffed out by many listeners. However, the results can still be incredibly good, and for a broad range of other material, not least orchestral solo spots with a touch of the *Schindler's List* spun-out melody, or more active writing akin to a Danse Macabre (minus the tritone double-stops of course), Stradivari Violin will do it, and can sound excellent. Short of employing a session player, it's one of the best options out there, currently. ▰▰▰

$ $199
W www.native-instruments.com

Felt Instruments
Blisko Collection

Sample Libraries

Felt Instruments' Blisko Collection offers a unique blend of sound and motion.

MARK NOWAKOWSKI

There is a particularly austere and engaging aesthetic to Felt Instruments products, which seemingly emerges straight from the soil of Polish culture. Trapped for years behind the Iron Curtain, the country managed to maintain a unique cultural and international profile instead of just melting into a larger Soviet soup. Despite strict censorship (or perhaps in spite of it), great composers emerged and with them entirely new ways of creating music. The film scores of Kilar and Preisner, the sonoristic experiments of Penderecki, and the new tonality of Henryk Gorecki are good examples. Through this time and into the new independence beginning in the 1990's, Poland had robust community music programs as well as excellent education in maths and later

computer science. In recent years her growing international creative profile has given rise to a revolution in video game making (such as the iconic *Witcher* series) as well as popular forays into streaming services such as Netflix. It was only a matter of time, therefore, before she gave us something of real consequence in the realm of virtual instruments.

Enter Felt Instruments, founded by composer and engineer Tomek Ślesicki in collaboration with the sensational genre-bending multi-instrumentalist Marta Zalewska (seriously, look up her YouTube videos). Their latest and most ambitious

effort is the Blisko series, comprised of a violin, viola and cello (available separately or as a suite). The Polish word Blisko actually means 'close', and the intense nearness of these particular instruments is effectively presented by expressive musicality and captured through the palpable warmth of vintage analogue gear to create a truly original effort. Curious composers should also check out Felt Instrument's encapsulation of the Hohner Guitaret, Jasno, which means 'bright', and super-soft upright piano, Lekko, which literally means 'light'. Also fun is the free Wolno (slowly) library of half-speed piano samples.

Shapes & Moments

What quickly becomes apparent is that Felt Instruments are not a Polish IKEA of samples with minimalist design and clever ethnic names; rather they offer expressive instruments with warm, sometimes gritty, and often evocative characteristics. The library is organised into two collections: the Shapes section contains sustained notes, col legno, unmeasured ricochet, and swells. The Moments section consists of varying textures with a strong sense of motion. All the instruments in the series sit in an attractive custom player whose interface
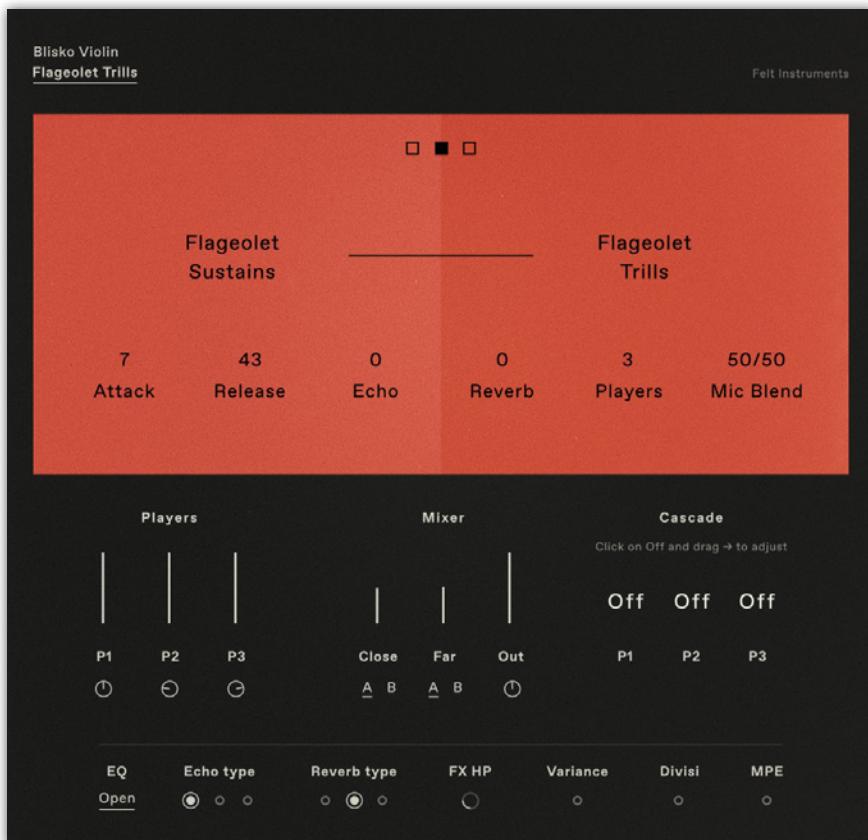
A Violin combo patch, allow the user to transition between harmonic sustains and harmonic trills.

Felt Instruments
Blisko Collection
£219

PROS
• Warm and characterful close-miked performances.
• Numerous shifting and aleatoric techniques which never sound like they are repeating.
• Simple and aesthetically pleasing custom-built interface.

CONS
• Lack of a dedicated sustain/legato instrument.

SUMMARY

A beautiful series of aleatoric and sound design tools for solo, duo and trio violin, viola and cello, allowing composers to create constantly shifting and morphing textures.
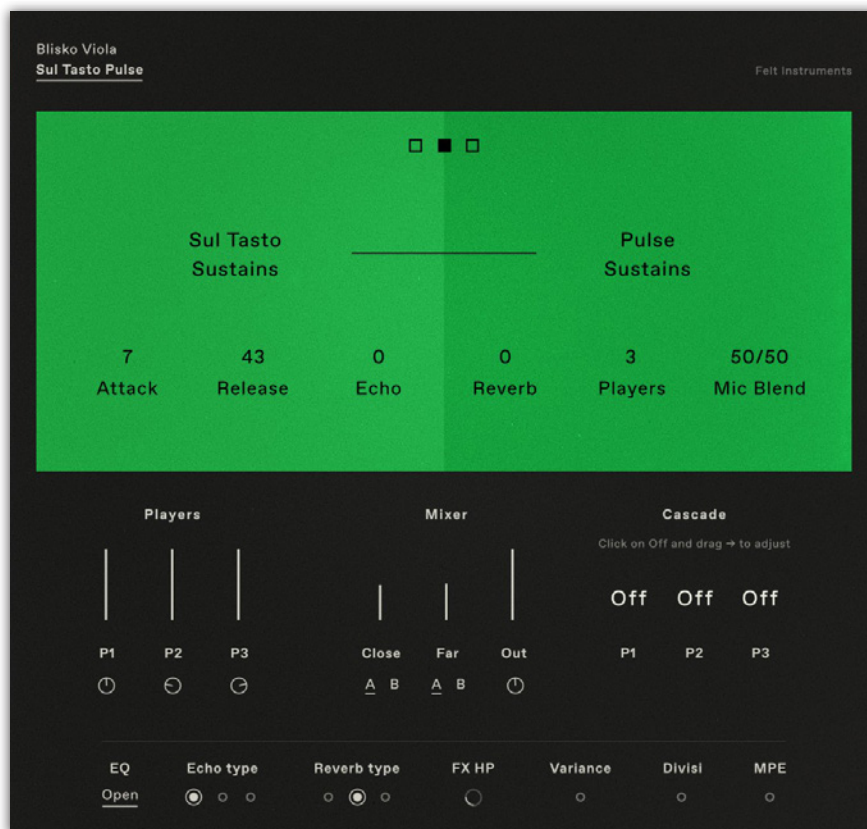
is certainly minimalist in inspiration and design. (The instrument is not standalone, however, and must sit within a DAW.)

Practically, this tends to translate to quick results in the creative realm. Upon loading your instrument, you can immediately affect the envelope, tape delay, reverb and mic blend (with two dry room and two close signals which can be blended) for each instrument while receiving attractive visual feedback. You can stretch and spread each instrument from a texture which covers between one and three players, effectively triggering the overdubs which provide for more variance in performed textures and also greater thickness. If you click on the 'Tweaks' tab at the bottom, you access a more detailed pane where echo and reverb types, the panning and volume of the individual players can be set, and a 'Cascade' option to delay the onset of the players is available: this is sometimes the secret to creating both great infinite sustains and more musical aleatoric performances. Also highly useful is the divisi switch. The EQ pane is similarly minimalistic and designed to resemble one of Tomek's Austrian EQ's. The three dials are joined by a Preamp button which further colours the sound.

Sound & Motion

As Ślesicki makes clear, his intent was not to create a one-stop comprehensive string library, but rather a unique tool for composers to add to their existing collections. Practically, this instrument is capable of adding additional life and motion to pad and aleatoric sequences while providing compelling new techniques and performances for existing string templates; I found it a particularly valuable addition to both my sketching and my string quartet templates. The dry nature of the recordings also makes it easy to match into existing ensemble string libraries. It's particularly fun to blend the performances and patches, spread to three players in the settings, into larger string sections, mixing esoteric techniques into a larger ensemble patchwork.

The first four patches in the library offer short performances. Ark offers a grainy, moody medium-length sustain with plenty of delicious bow noise, which retains its character even when the reverb setting is way up. Rise offers a short and more aggressive articulation, while Col Legno does exactly what you would expect at first glance, though some tweaking leads to more interesting results. Try, for instance, to

A Blisko Viola combo patch allows users to transition between Sul Tasto sustains and a pulsing figuration.

spread to three players, change the value of the cascade setting, and turn up your reverb while changing to the third type: the Col Legno then takes on a very different characteristic due to how its short sustain interacts with the new reverberant settings; the ricochet setting works similarly.

Beginning with Sul Tasto Pulse, the second line of available presets contains the 'Eternal' sustains, and often lets the user crossfade between two and three performance settings by using the mod wheel. Users will notice that three small squares have appeared near the top of the interface: this allows you to toggle between a single articulation type, a mod wheel crossfade between two types of articulations selectable by the user, and a crossfade between three chosen articulations. This is where the real magic of this clever little collection takes place. As always, the divisi switches allows you to have individual players play the techniques, or groups of three (to quickly bring you to a larger section density). The third and final line of sounds are more pointillistic in nature though also sustained indefinitely.

If there is one complaint to level against Blisko, it is that the sounds are so very nice that the user naturally desires a few more articulations. One or two more standard sustain patches or some manner

of repeat or re-bow setting would add even deeper functionality to an already useful product. As Ślesicki has updated the instrument numerous times even during the writing of this article, there is no reason to believe that this labour of love will not continue to grow along with his company's other efforts.

The website samples of this instrument are entirely representative of its capabilities, so interested composers and sound designers can confidently audition capabilities there.

Conclusion

Felt Instruments punch well above their weight with this useful tool. Yet perhaps the most telling aspect of whether a VST transcends function and enters into the realm of art is whether or not it is capable of inspiring new music in its own right. For my money, the Blisko collection is a goldmine of moody inspiration and pathos-driven performance, and one which certainly will enrich the creative pallet and digital arsenal of many composers. ◢◢◢

$ £219 or £79 each.
W www.feltinstruments.com

Steinberg
Cubase 11

Digital Audio Workstation Software

Steinberg turn the dial up to 11 with a stonking new version of Cubase.

JOHN WALDEN

While death and taxes may still be the trump cards in the game of certainty, the inevitability of a tempting upgrade to your favourite music production software can't come far behind. For many years, Steinberg have been very consistent on this front, with annual upgrades to Cubase released towards the year end. Cubase 11 — in Pro, Artist and Elements versions — is therefore now with us.

These regular annual upgrade cycles bring out different responses in users. Those happy with their current version can, of course, choose to stick but, when presented with a long list of improvements and new features, the temptation to twist can be high. So, just what are Steinberg adding in Cubase 11 that might entice

existing users to upgrade, or tempt potential new users to take the plunge?

On The Shoulders Of Giants

The current DAW/sequencer marketplace includes many excellent choices, and Cubase is undoubtedly one of the 'giants' amongst this product group, with a long history, a large user base, and a comprehensive feature set. While v11 brings all sorts of worthwhile new developments, if you are part of that large user-base and Cubase helps put food on your table, you can breathe easy; moving from any recent (10 or 10.5, for example) version to Cubase 11 is a smooth and painless transition.

That said, on paper, for users of the Pro version, the list of 'new and improved' in this upgrade is impressive. And while not all of these new features trickle down to the Artist and Elements versions, in both cases, there are some nice surprises. For example, Elements now includes side-chaining options and Artist now includes support for VariAudio 3, ARA2 and unlimited audio, MIDI or instrument tracks. So, what's new? Let's dig in...

My Superpower Is...

While there is truth in the old adage that if it sounds right, it is right, a visual representation of your audio can offer all sorts of helpful insights. Acquiring top-notch audio visualisation skills might, therefore, come a close second to 'golden ears' on your list of desirable superpowers as a budding mixing or mastering superhero. Cubase 11's new SuperVision plug-in provides an excellent starting point in that regard.

SuperVision is a single plug-in, which can be placed as an insert anywhere it's required to visualise your audio signals, but is actually an assemblage of modules that the user can configure and customise. The 18 available modules

fall into a number of categories: Signal, Spectral Domain, Phase, Spatial Domain, Waveform and Other. Many of these are relatively straightforward, such as the Level module within the Signal category, or the Spectrum Bar (a frequency-based histogram) within the Spectral Domain category. However, the range of options is impressive and a few of the modules are worth further comment.

For example, the Signal category includes a well-featured Loudness module that follows the EBU R128 reference. The module displays true peak, integrated loudness, short-term loudness, momentary maximum and loudness range and, amongst other configurable options, is switchable between LU or LUFS units. In the post-loudness-wars world of streaming, whatever your specific loudness target, SuperVision's Loudness module will assist you in hitting it.

Alongside the very useful, but fairly conventional, Spectrum Curve (which,

---

Steinberg Cubase 11
$552

PROS
- Frequency 2 and SuperVision justify the cost of the upgrade on their own.
- An extensive selection of new and enhanced features that will have very broad appeal.
- Impressive new audio export options.
- Scale Assistant is an absolute boon for non-keyboard players.

CONS
- No new virtual instruments to get excited about.
- Er... nope, can't think of anything else.

SUMMARY
Cubase Pro 11 delivers an impressive array of new and enhanced features that will have a very broad appeal. Tremendous value for those upgrading.

---

interestingly, does offer a masking feature that can compare a main signal with that of a side-chain input) and Spectrum Bar, the Spectral Domain modules also include Spectrum Intensity, Spectrogram and Chromograph options. The latter provides an intriguing display which shows the energy of your audio based upon the 12 half-steps of the equal-tempered scale.

The Phase modules provide various ways to visualise the correlation between the left and right sides of a stereo signal. The Phasescope, Panorama and Correlation modules are familiar, but I particularly liked the Multipanorama module. This is frequency-based and particularly useful for checking out just how 'stereo' your low end is. It could make a good companion to the new spatial imaging plug-in described below. The Spatial Domain modules — Surround and Ambisonics — offer useful level and channel correlation visualisation for those working in multichannel formats also.

With the ability to fully customise the choice, position and size of up to nine modules within a single instance of SuperVision, and to save your own layout presets, this is a powerful and flexible visualisation system. While they do differ in many details, an obvious comparison would be with iZotope's powerful Insight 2, which also provides a modular visualisation system. Purchased individually, Insight is priced at $199. SuperVision now provides a very credible alternative for Cubase Pro and Artist users, who'll get it with the Cubase 11 upgrade.

Squeeze, Spread, Squash

Cubase 11 delivers three notable plug-in developments. For Pro users only, Frequency, the most powerful of the stock EQ plug-ins, has been enhanced to Frequency 2. New to both Pro and Artist is Imager, a multiband stereo imaging plug-in. And new to Pro, Artist

and Elements is Squasher, a three-band dynamics processor that combines both downwards compression of peaks and upwards compression of quieter parts of your signal.

Frequency 2 adds dynamic EQ capability to an already powerful feature set and also provides support for multiple side-chain inputs (a capability that's part of the VST3 specification). For any of the eight bands set to low shelf, high shelf or peak filter types, dynamic mode can be engaged, with a full suite of controls to configure how the EQ band responds to the dynamics of the audio input within that frequency range. The dynamic EQ options are available whether an EQ band is in standard stereo mode or being used in Mid/Sides or left/right modes, in which case you get separate sets of controls for both channels.

The new multi-side-chain support opens up all sorts of interesting possibilities. For example, if you have

a group track acting as a bus for all your guitars or synths, you could duck the low frequencies of this from a bass or kick-drum side-chain input, while simultaneously ducking a dB or three somewhere in the midrange based upon a side-chain from your vocals. If side-chaining is part of your toolset to ensure key rhythmic or melodic elements remain present and intelligible within a busy mix, Frequency 2 will be a powerful addition. An obvious point of comparison is FabFilter's highly regarded Pro-Q3 plug-in, and while the two do have their differences, Frequency 2 provides much of the same functionality and workflow for Cubase Pro users.

Squasher provides up to three bands of dynamic control, offering both conventional downwards compression (to control loud signal levels above a threshold) and upwards compression (to increase the volume of quieter portions of the audio signal below a second threshold). Allowing you to control the amount of compression in both directions, this is a very flexible tool for managing the dynamics of any audio signal.

Steinberg suggest Squasher is aimed primarily at EDM producers and, with each bands each having variable drive and a side-chain input, it's easy to see how you could push a contemporary electronic dance mix to within an inch of its dynamic life should you so wish. However, I think that might undersell its utility. For example, used in single-band configuration, the combination of upward and downward compression is similar in principle to the MV2 plug-in from Waves. This is a great tool for quickly evening out the dynamics of a vocal or bass and Squasher can now fulfil the same role for users of all Cubase versions.

It also works well to bring out lower-level details in a sound such as ghost notes on a snare mic or rhythmical muted strums within a guitar part. In short, this is a very useful addition to the Cubase plug-in suite, but I'd qualify that with a warning that you can just as easily destroy a mix as enhance it; use with suitable restraint!

Unmixing Magic

Steinberg have recently released a major upgrade to SpectraLayers Pro (a full review of SLP7 will appear in *SOS* soon) but, for Pro and Artist users, Cubase 11 bundles SpectraLayers One, a compact version of the powerful spectral editor. And thanks to the ARA2 Extension system, these spectral audio editing capabilities can be opened for a selected audio clip available directly within the lower zone of the Cubase Project window.

SL One offers a more modest feature set than the SL Pro and SL Elements

Frequency 2 adds dynamic EQ and multiple side-chain support to Cubase Pro's flagship EQ plug-in.

Test Spec

Cubase Pro 11.0.0
Apple iMac running OS 10.15.7, 3.5 GHz Intel Core i7, 32GB RAM.

versions. However, perhaps the most eye-catching SLP7 feature included in SL One is a streamlined version of the almost magical AI-based stem unmixing. As in SL Elements, One is limited to the Unmix Vocal option, taking a fully mixed audio track and separating it into two layers: 'vocal' and 'everything else'. But for those looking to do remixes or mashups, that's exactly what's needed. It requires just a couple of mouse clicks to work its magic and the results are, well, remarkable. When played back together, the two layers sum to the original audio, but you can also choose to solo or mute one to get either just the vocal or just the instrumental bed. The latter pretty much always sounds very (very!) good, and if you need to simply rebalance the vocal against the instrumental bed by a dB or two, SL One will do that in a very transparent fashion.

The quality of the soloed vocal will vary depending upon factors such as how busy the original mix is, the amount of reverb/delay applied to the original vocal and the presence of prominent backing/harmony vocals. However, if you have ever tried to extract a vocal from a full mix by any other means, you will appreciate just how good a job SpectraLayers is doing. And if you do need to clean things up further, significantly, SL One includes the new Cut To Layer Below option so, as you refine the separation by cutting unwanted elements from the vocal layer, they can automatically be preserved on a further layer (or layers) to ensure the process remains fully non-destructive.

A Slice Of Something Nice

Available in all Cubase editions, Sample Track 2 retains the easy-to-use workflow of the original but adds some creative new features. The highlight amongst these are the slicing options; although loop-slicing is a well-trodden path in most mainstream DAWs, having the capability embedded directly within the Sample Track provides a very welcome creative and workflow boost.

Rather than just allowing you to toggle AudioWarp mode on or off, the Playback panel (now with a blue header strip) now includes three sub-panels: Normal, AudioWarp and Slice. AudioWarp works as before, allowing you to maintain a playback speed while shifting the pitch of the sample playback across the MIDI note range. However, the new Normal sub-panel now allows you to select between different playback qualities, depending upon just how pristine you want your sample playback to be. This includes a cool Vintage option to simulate old-school sampler technology by reducing the bit depth or the turntable speed.

As might be expected, Slice mode divides your sample into slices using automatic transient detection, manual slicing, slice to a grid or some combination of these. The slices are mapped across the MIDI keyboard. Using the small MIDI connector icon, you can drag into the Sample Track a MIDI clip that will replay the sample slices in sequence to recreate the original loop. This can, of course, then be edited to create variations of the original loop or entirely new loops. Like all loop-slicing tools, the process is designed with drum loops in mind but it works well with any material that contains obvious »

Frequency 2 includes a single-band view for more detailed control and access to the side-chain options.

transients; classic EDM vocal chops, for example, are a breeze.

The Sampler Track functionality now also includes two LFOs. These can be used to modulate key parameters in Pitch, Filter and Amp panels, while clicking on the Mod button for any of these panels to open its envelope editing also displays the various LFO controls. While it's perhaps not a match for the modulation options found in many of Cubase's virtual instruments, it does provide some very creative sound-shifting possibilities.

The Mono button on the Sample Track's toolbar strip — which forces sample playback to one note at a time — now has its own dedicated Legato button. Once engaged, if you play a second note while holding the first, the pitch alters but sample and envelope playback continue without retriggering. You can combine this with the Glide option and the new Fing button (short for Finger, this forces a pitch glide only when you play legato) in the Pitch panel. You can create some very cool retro synth-bass or melody lines with suitable sample starting points.

Oh, and if you want some fresh audio ammunition to explore the new Sample Track 2 features, Cubase 11 also offers six new sound sets: Bloom, Noir, Night Call Synthwave, Hard Knocks, Lofi Dreams and Dance Floor Tech House. These are included for all Cubase versions.

Theoretical Advice

There is lots to be excited about in the new features already described but, if your piano keyboard skills are modest (like mine), or you have yet to pass Music Theory 101 (cough!), then the new Scale Assistant within the MIDI Key Editor — present in Pro, Artist and Elements — may well be the absolute highlight of the v11 release. The concept is simple: if you ask nicely, Cubase will pitch-correct your MIDI note data to fit the chosen key/scale combination of your project. It can do this to existing MIDI parts or to 'live' MIDI input. It can follow a scale or some combination of the scale and associated Chord Track. And, when you edit MIDI data, it can snap to notes within the scale or chord. In short, no more duff notes!

At a practical level, it is worth noting that the Scale Assistant is only active within the Key Editor. You therefore have to have an existing MIDI clip (with notes you already recorded or an empty clip you have created) selected and open within the Key Editor window. From here, you can then open the new Scale Assistant tab in the Key Editor's Inspector panel. When using the Scale Assistant, it is also useful to select the Scale/Chords option from the Event Colours entry on the Key Editor's toolbar. Notes will then be colour-coded to indicate whether they lie in or out of the chosen key/scale.

Squasher provides both downwards and upwards compression across three frequency bands and with support for multiple side-chain inputs.
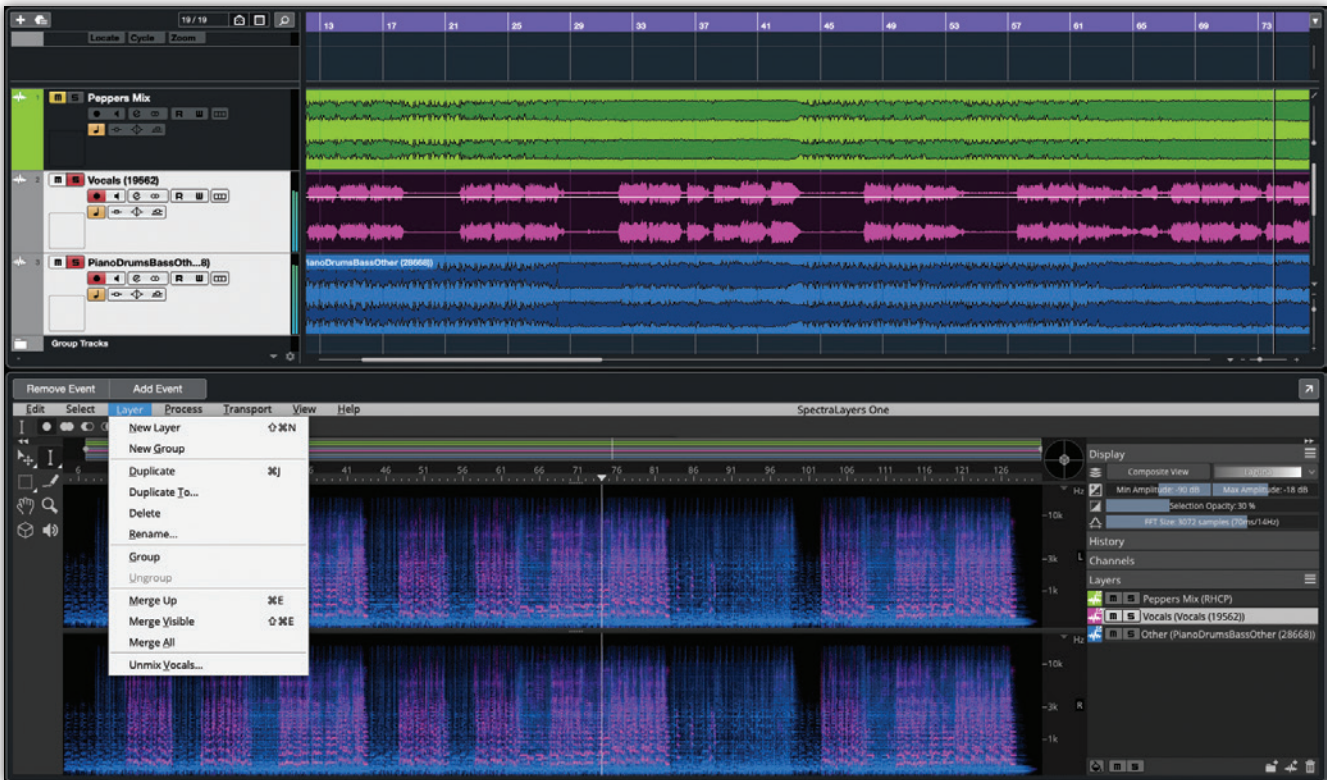
MASCHINE +

NO COMPUTER NECESSARY.

SpectraLayers One adds the magic of vocal unmixing for Pro and Artist users.

The Scale Assistant panel includes a number of options. You can choose to use either the scale specified within the panel or any scale specified in the Chord Track (sensibly, this is an either/or selection). If you opt for Use Editor Scale, you can then select a key/scale combination from a well-populated menu, with the option to add your own. If you have an existing set of notes within the clip, but are unsure of the key/scale combination, the Scale Assistant can also offer you some suggestions based on all the notes or just a selection, although if your notes just happen to be an assemblage of randomness, it is possible to get 'zero' suggestions.

The three tick boxes define the level of music theory support then offered. Show Scale Note Guides adjusts the pattern of light and dark grey horizontal bands that run across the MIDI note area from the default white-note/black-note pattern to an in-key/out-of-key pattern to provide a visual guide for note selection. Snap Pitch Editing takes the help a step further so that, when you edit MIDI notes, they snap to pitches within the correct key/scale. Finally — and this is the best bit for those with limited keyboard skills — Snap Live Input will pitch-correct your MIDI note entry as you play. This works well, but

do note that as the Scale Assistant is a Key Editor process only, you need to engage the Record In Editor button (far left of the Key Editor toolbar) to make this active. Finally, Quantize Pitches can be used to correct any notes that lie outside the key/scale in existing MIDI.

I can see this new feature finding a lot of friends amongst non-keyboard players, although it perhaps takes a little experimentation to find your ideal workflow. In my own experimentation, entering chords from the Chord Pads and then applying some Scale Assistant-based editing of the MIDI data made it very easy to explore different chord voicings or note tensions, while keeping everything in key. Used for melodic or bass lines, however, the Scale Assistant is impressive on its own; simply record a suitable take (with or without live pitch-correction) and then cycle playback while you fine-tune the melody with Snap Pitch Editing activated.

Finally, it's worth noting that if you select a melodic line and shift a copy of it by a specific musical interval using the first note of the melody as a guide, with Snap Pitch Editing engaged, the notes move intelligently rather than all just moving the same interval (for example, three semitones). Music theory skills or

not, this will be an absolute boon when creating harmony lines for things like string parts.

Round The Bend

While on the subject of the Key Editor, Cubase's MIDI editing gets further refinements with new CC drawing options added. As has been the case for some time with automation data on audio tracks, you now have options for creating smooth curves between two MIDI CC data nodes. This works with any MIDI CC lane within the standard MIDI editors. One further nice touch is that, when editing pitch-bend data, you also get the option to snap the data to a semitone-based grid. While some might argue this feature is a long-overdue addition, it's great to see and it's available in Pro, Artist and Elements.

Export Ecstasy

Whether for personal archive needs, to send to a mix engineer, or to give a client maximum flexibility down the line, exporting more than just a final stereo mix from a project is often a necessity. However, when that might consist of bus-based stems and full suites of audio and instrument tracks (both with and without any included channel-based effects), it can also be a chore that requires patience and concentration.

━━ The display of Global tracks within the Key Editor will be very helpful for composers when synchronising musical content to specific visual events in their film or TV project.

›› For Cubase Pro users, this is already made somewhat easier by the more comprehensive Export Audio Mixdown dialogue in the Pro version. It remains, however, a somewhat tedious task, and mistakes can easily be made.

However, Cubase Pro 11 introduces further key additions to the Export Audio Mixdown dialogue that can make the process considerably easier to manage, no matter how complex the export specifications. First, there's a new Sync Selection To Channel/Track Selection button. Engage this, and tracks selected within the Project window will be automatically selected for audio export when the Export Audio Mixdown dialogue is opened. Second, multiple different export jobs can now be queued so that they are automatically carried out one after the other. Once one selection has been made and added to the new Export Queue list, you can then make further selections and create additional jobs as required. You can also open an existing job within the queue and update its contents by adding more selected tracks or deleting some that are already included.

This is both a powerful and flexible system. Separate jobs can focus on different elements, so you might create a job to export just the main stereo output (your final mix), a job for an MP3 version of the same mix, a job containing all your group tracks (submix busses) and a job that exports all the individual audio and instrument tracks as audio (perhaps for an external mix engineer or as a future-proof backup). What's more, each track selection can be configured with different elements of their respective processing chain included. So, if you want to export all your audio tracks both with, and then without, their channel and insert effects processing, for example, that's easy to configure into two separate jobs.

Queuing jobs in this way allows you to check you have everything organised before starting to actually export the required files. Usefully, the jobs queue is saved with your project so you can build the list of jobs over multiple sessions if needed. Once your jobs are ready, you simply press the
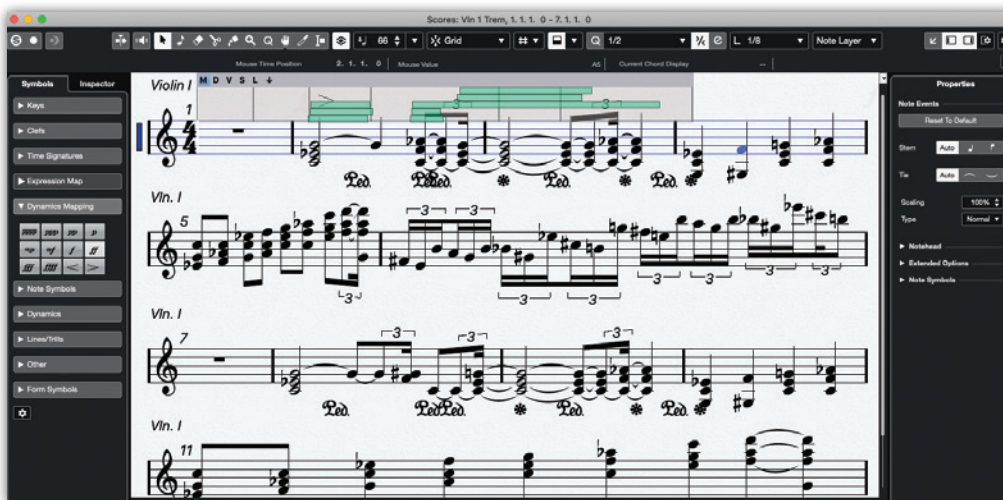
━━ The Score Editor has a new context-sensitive Properties panel and a new note-editing overlay.

Start Queue Export button and, if there is a lot of stuff in the queue, you then let Cubase do its thing while you make a hot beverage of your choice... although the first few times you use the queue feature, it's simply a joy to watch while wearing a big smile. Oh, and if you still discover that you have missed something, you can simply make the necessary changes or additions to the jobs within the queue and execute the process again.

A further useful detail is that all exported files can be placed within a dedicated Mixdown folder within the Project folder rather than the Project's Audio folder. This is a sensible option, keeping your exported audio separate from the project's original audio recordings, and makes managing your various audio media much more straightforward. However, it would be even better if you had the option to put exported files into their own sub-folder within this Mixdown folder based upon the job number. You could do this manually when configuring each job, but an automatic option would be a further nice

addition at some stage. Even so, job-based batch export is a great addition for Pro users with complex export requirements.

Compose Yourself

The new export options will undoubtedly appeal to media composers required to deliver multiple stems for a film or TV project, and there are a couple of other 'new in 11' options that will also particularly appeal to that part of the Cubase user base. First, if you regularly float the Key Editor to make it larger than the lower zone might allow, you now have the option to toggle on/off the display of key global tracks within your project — Markers, Chord Track, Ruler, Arranger and (yay!) Video — at the top of the Key Editor. This can be done from the new Global Tracks tab that appears in the Key Editor's Inspector panel when it is floated. The system is simple to use and very effective. The ability to see markers (showing musical hitpoints), timecode and a video thumbnail while undertaking precise MIDI editing will be a big help to anyone synchronising their musical compositions to picture.

Second, Steinberg have also made some refinements within the Score Editor for Pro users. For example, as the Score Editor now supports the SMuFL font format, some of the more attractive fonts available in Dorico (such as Bravura) are now available in Cubase. To speed workflow, there is now also a new Properties tab (available in the left panel of the Score Editor) that is context-sensitive; you get appropriate options displayed within the tab as you select particular elements within the score. Finally, there is now a new note-editing overlay panel that can be toggled on or off and allows you to refine your editing choices within the actual score. You can, of course, edit note length, velocities and pitch here, and the changes are automatically applied to the underlying MIDI data. However, if you wish simply to improve the intelligibility of the score for sight reading, you can adjust note lengths within the score but without changing the MIDI note length. These changes are obviously useful for those who need to generate printed scores for »

» musicians, and might mean that some smaller scoring jobs can be done within Cubase itself, with no need to resort to a full-blown scoring package such as Dorico.

In Heaven With Eleven?

Given just how mature the DAW/ sequencer marketplace now is, it's difficult to see how any major DAW update might be thought of as revolutionary rather than evolutionary. Indeed, for loyal users of a particular DAW/sequencer, the ideal combination might be a new feature or two that give a nod to revolution, mixed with a good dollop of smooth evolution. Existing users get to keep their well-established workflows intact (with some useful new possibilities) and the expanded feature base, with a few highlights, can tempt potential new users. So, does the Cubase Pro 11 update achieve that ideal balance of mini-revolution and solid evolution? And does the upgrade price for existing Pro users deliver value for money?

Let's start with the second of these questions. The new features add up to a serious collection of additions and significant refinements, and I think it's difficult not to be impressed by the value for money on offer. Even if you consider just the combination of SuperVision, Imager, Frequency 2 and Squasher, there is plenty of value in these new or revised plug-ins to more than justify the price of the upgrade. You are, as I mentioned earlier, getting functionality that compares with premium third-party plug-ins such as iZotope's Insight 2 and Imager, FabFilter's Q3 and Waves' MV2; and these are, of course, only part of what you get when moving up to Pro 11. With the possible exception of the Score Editor improvements, I think the majority of the new and enhanced features fall into the mainstream, and a large proportion of the user base is likely to see a benefit. I'd suggest this is a very impressive deal for existing users.

Is there anything revolutionary? Well, there are elements that revolutionise the way particular tasks can be approached. For example, with Frequency 2, Imager and Squasher now added to the existing multiband processing options provided by the Multiband Compressor, Multiband EnvelopeShaper, Multiband Expander and Quadrafuzz 2, and with the new SuperVision providing sophisticated
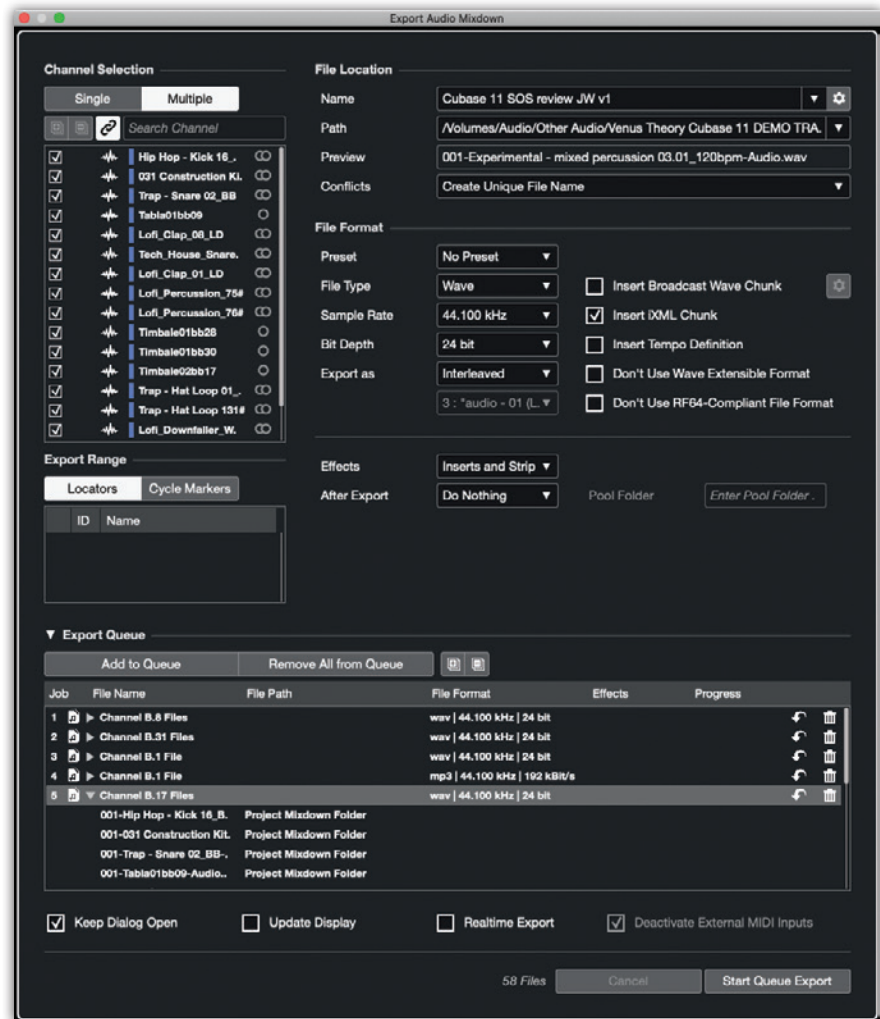
Synchronisation of track selections between the Project window and the Export Audio Mixdown, and the new Export Queue system, make organising complex export tasks much easier in Cubase Pro 11.

visual analysis and loudness monitoring, Cubase Pro 11 has a very powerful set of tools for audio mastering. Yes, if I'm doing DIY mastering on a collection of tracks, I'd prefer the Audio Montage environment offered by Wavelab but, if you are mastering a track or two for your latest single before it hits all the streaming services, Cubase Pro 11 has all the tools you need, provided you have the skills to use them.

There is a similar 'workflow revolution' for media composers in terms of the new job-queuing feature within the Export Audio Mixdown dialogue and the Global Track display within the MIDI editors. These are evolutionary changes that have the potential to revolutionise the workflow for these specific tasks. Equally, if your piano keyboard skills suck, the evolution of the Key Editor with the new Scale Assistant may well revolutionise your ability to create harmonically correct MIDI parts.

Cubase Pro 10.5 was already a massively powerful and feature-rich musical production environment; it therefore provided some pretty broad shoulders upon which the new features of version 11 can stand. However, as a long-standing Cubase Pro user — and one who has followed many iterations of the annual upgrade cycle — I think this upgrade is one of the most impressive I've seen. It delivers a broad sweep of genuinely useful new and enhanced features that will appeal to a large number of existing users and, along the way, delivers some very appealing mini revolutions to workflow in specific areas. This is a truly polished upgrade that delivers exceptional value for money. ◢◢◢

$ Cubase Pro 11 $552; upgrades from $99.99; Artist 11 $302; Elements 11 $99.99.

W www.steinberg.net

Success In The Studio Business

Three Tips For Getting Ahead

A Grammy-winning engineer shares his secrets.

DOM MORLEY

Firstly, I feel I should apologise for the clickbait headline, but I promise this is more than an attempt at 'attention engineering'! Whilst everyone's route to success in the studio world is different, there are a few tactics that I've used, and which I've seen other people use, to get ahead. These tips are definitely 'tried and tested', but I'll stop short of 'guaranteed' for legal reasons.

Be Thorough

The first obstacle that's rarely talked about is the surprising lack of transferable skills between the jobs of assistant, engineer and producer. There are a few things that you carry with you, but knowing the details of what's in the mic cupboard won't teach you how to mix, and being a good mixer won't teach you how to coax a great performance from a fragile singer having a bad day. One thing that is consistently transferable though, is being thorough.

This advice was given to me early in my career and I've always found it to be useful. Whatever the task in front of you, approach it with the mindset of being thorough and you will do the job with professionalism, and do it well. These things get noticed, and when you are getting noticed then you start moving ahead. Early in my own career I moved from a small regional studio where I was

an assistant who was starting to get engineering gigs to a big London studio where I was at the back of a daunting queue of experienced assistants and engineers. My own engineering opportunities disappeared over the horizon! So, I got to work doing the tasks that I was given really well. If all I was doing was making drinks, then I would remember people's orders so when they turned up next time their favourite style of coffee would be waiting for them. Being a specialist in making hot chocolate and ice tea also made me very popular! Slowly this attention to detail gets you noticed, and when it's time to trust someone with a more important task, you're the one who is trusted.

This applies to engineering and production too. The engineer who always delivers a great recording is likely to be asked to mix a track when the budget doesn't stretch to fit in the usual Big Budget mix engineer. And the mixer who goes the extra mile to really get the best out of a track might be asked to produce the next record. It all comes down to your enthusiasm to get the task done as well as possible. When that shines through, it becomes infectious and attracts bigger and better things.

Keep Learning

Studio work is obviously as technical as it is creative, so it should go without saying that you never stop learning — both your

The Author

Producer, mixer and recording engineer Dom Morley has multiple Grammy awards under his belt. Among his credits are Amy Winehouse's *Back To Black* and Mark Ronson's *Version*, plus singles by Adele and Morrissey, Sting's 5.1 surround release *Live At Durham Cathedral*, and many more. Dom recently appeared on the *SOS* 'Recording & Mixing Vocals' podcast, which you can hear at https://sosm.ag/vocals-podcast.

craft and the new technologies associated with it. Being at the forefront of new technologies can be a big help in moving forward as well. If you are the person who knows how to make a certain bit of software or hardware work when no-one else does, then you can become in demand very quickly. This is often a small window of time as the rest of the studio community catches up, but it can be enough to push your career forward, especially if you're also demonstrating everything in the 'Be Thorough' paragraph above, as well as the paragraph below...

Be A 'Cool Hang'

I guess I need to apologise again for that heading, but a 'cool hang' is how it was explained to me, it sums it up well, and I'm guessing you know what I mean. Studio sessions can be long, often running for many days in a row, and they can get quite intense. Bear in mind that people hire the best person for a job that they enjoy working with, so making the effort to be a really nice person to hang around with is really going to be critical to your career. This gets into the area of 'vibe' — something undefined that we are all somehow supposed to understand — but I'll give a few examples of what works.

Consider things from the artist's point of view. It's stressful for an artist to be finally recording and/or mixing tracks that they have been working on for a very long time and those stresses can play out in different ways depending on the person. Start with an assumption that that's what is happening.

'Diva' attitudes are born from insecurity. It's the job of every person in the studio to make the artist feel secure and comfortable, so if you're getting some of this stress then think what you can do to make the artist feel more confident and supported.

Never take diva behaviour personally, because it really isn't personal — and keep doing your job with enthusiasm. Eventually the tension decreases, and life gets much easier for everyone.

It's really that word 'enthusiasm' that's the big one here. If you can make sure you are always injecting that into your work, then all of the above advice should fall into place without a great deal of extra thought. Best of luck! ∎∎∎

LCR Panning

A Mixing State Of Mind

LCR panning isn't so much a technique as a mindset
that can lead you to more satisfying mixes.

MATT HOUGHTON

There can be a huge gulf between what trained and untrained ears hear when they listen to a track. Your average listener isn't thinking "That guitar's a bit edgy," or "Isn't the conga fighting a bit with the floor tom?" Their reactions are much more instinctual and basic. Music either provokes an emotional response in people, or it doesn't. It's all too easy to lose sight of that basic fact when you're mixing. Yes, a focus on detail can be important, but if you focus on the wrong details you'll end up making the track sound fussy and confused, rather than clear and powerful. LCR panning can help you avoid falling into that trap. It isn't a recipe that you can follow to make any track sound good, or a tool that you can use to correct problems. Rather, it's a discipline that can help you to really focus on what matters in an arrangement and a mix.

Discipline

The idea is simple: don't use intermediate pan positions. Any given mono source can be panned hard left, hard right or straight down the middle. Nowhere else. Of course, at one time, engineers had no other option: early stereo consoles had three-position switches to route signals to the left channel, the right channel or both channels of the master stereo bus.

But today, pretty much anyone using a mixer or DAW software has access to continuous pan controls. If you have a lot of sources and you mix in stereo, then, there can be a strong temptation to use intermediate pan positions to make them all audible. Yet, a likely consequence of doing that is that you end up overloading the listener's perception. On a good day, all those sources might lock together to create an impressive wall of sound, but often it just ends up sounding like a directionless mess.

Technically, a mono mix is an LCR mix, and in many ways LCR can be considered an extension of the idea of mixing in mono: it's almost like treating the left speaker, the right speaker and the phantom centre as three semi-independent mono mixes. Mixing in mono forces you to make a decision as to which of the sources you think are really important, simply because there is no way that all of them can be equally prominent in a mono mix. It forces you to treat those key sources in such a way that they fit around each other, whether that's through fader moves to bring them in and out, arrangement tweaks, dynamic variation or ensuring that they all occupy different frequency ranges.

Although mixing in mono is a fantastic discipline, few of us would actually

want to submit mono masters unless specifically requested by the artist or label. Mono-compatibility remains important, but most people are listening in stereo, so naturally we want to take advantage of the additional possibilities offered by two speakers. LCR panning is a way of doing this whilst still adhering to the core disciplines associated with mixing in mono.

LCR panning is sometimes called 'cardinal points panning', the inference being that the extremes and the centre of the mix are the points that really matter. Rather than thinking of the stereo field as a huge playground in which your sources can frolic at will, the idea is that if you get the balance right at those three points, the spaces in between them should take care of themselves.

Is it easier to mix in LCR? Not necessarily. The same limitations and the same considerations apply as usual, and the whole mix still needs to work when it's auditioned in mono. In fact, when you first try it, it can seem harder. But in my experience that's generally because LCR is an excellent tool for revealing more fundamental problems on the project: if a track is difficult to mix in LCR, that's very often because the underlying arrangement or tracking is flawed in some way. Perhaps the greatest strength of LCR panning is that it guides you to make the kinds of decisions that matter to untrained ears — to your audience, in other words. Thus, it can increase your chances of crafting a successful and satisfying mix.

Producer Terry Manning proposed the 'cardinal points law' technique for panning as a means of ensuring that the stereo mix doesn't become too congested.

Before you fire up your DAW and get going, though, it's worth running through some issues that might lead to confusion or, potentially, disappointment for people mixing in LCR for the first time. Happily, they're not so much problems as characteristics: the idea of exploring them here is that if you encounter them you'll know what's going on and be able to make the appropriate decisions.

Separation Anxiety

I don't want to give you the impression that all sounds in a mix inherently need to be perceived as separate — sometimes the magic lies precisely in those moments when sounds combine to create an intriguing texture that the listener can't pinpoint. But, typically, in every mix we want to prevent certain sounds obscuring or changing the tonality of others.

One charge sometimes laid against LCR panning is that the hard panning creates a false sense of separation that can cause problems when you hear the mix in mono. But while there can be no doubt that LCR panning contributes to a sense of spaciousness in a stereo mix — there is literally more space in the phantom centre, and the width of the mix »
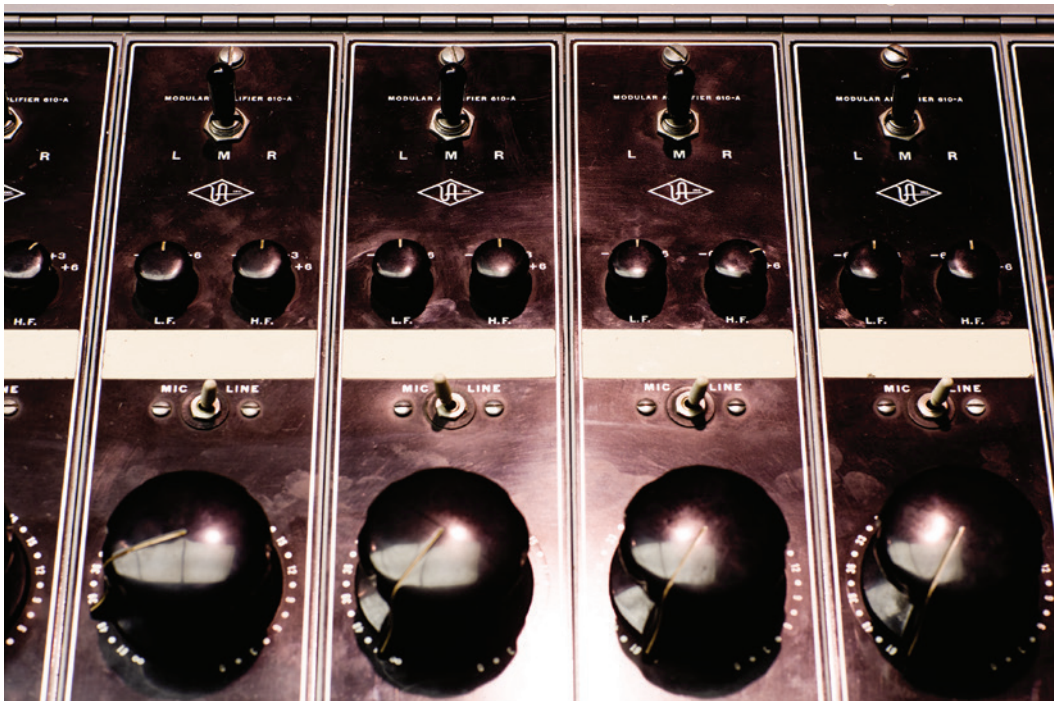
Early stereo mixing consoles such as Bill Putnam's Universal Audio 610, pictured here at Valentine studios, didn't offer continuous pan controls, just a switch to route signals to the left, right or both channels of the stereo bus.

when you switch to listening in mono.

Especially where you have very few elements in a production, there's a risk with purist LCR panning decisions that the mix can feel a little off-balance at times. Personally, I quite like bold stereo moves, where one part comes in very obviously on one side and another might join in on the opposite 'wing' a few bars later.
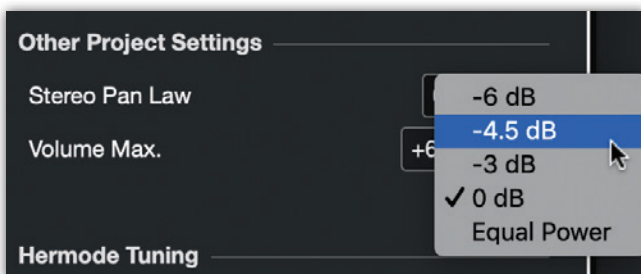
But it doesn't always work, and you'll rarely want the overall left-right balance to be skewed for the full duration of a song, so sometimes when LCR panning you'll want to find a way to counterbalance a hard-panned part.

When you want to put a part on one side but doing so creates a sense of imbalance that you find annoying, you'd usually pan something else opposite to counterbalance it. In that situation, if there are no other parts available then it's really an arrangement issue: you could probably solve it by writing a new part for the other side or perhaps having a new instrument double the existing part there. But, assuming that's not an option, there are also viable mixing techniques.

For example, you could try creating a fake double-track and use that on the other side; you'll lose the sense of the part being panned one side or the other that way, but you'll still retain the sense of width and space around the centrally panned sounds. Often better, if you want to retain the sense of the sound being on one side, is to send the sound to a mono reverb or delay effect, whose return channel is panned opposite: you'll still perceive the sound as emanating from the point at which you hear it first, but the dry source will seem less like a sore thumb.

Balancing Act #2

Whenever you sum a stereo mix to mono, even if you've created perfect separation

emphasises that even more — it's a false charge, since it assumes that the whole point of LCR panning is that it creates separation through panning. That's not the aim, and neither would it be a good tactic.

Here's a common example that explains why: the big, wide wall of distorted guitars you hear in so many modern rock and metal tracks. It can sound truly impressive when monitoring in stereo if you simply bring up the faders and opposition-pan (hard-left and hard-right) your double- or quad-tracked guitar parts, and leave the electric bass in the centre. But the wide-panned guitars and centrally panned bass will always have overlapping frequencies, and if you listen in mono you'll almost certainly find that you want to high-pass filter the guitar parts to prevent their low end from smearing the bass (it needn't be aggressive: a 12dB per octave filter at around 140-150 Hz usually works fine). You might also want to investigate higher up the spectrum, to see if the guitars compromise the note definition in the bass part, particularly if they do so

intermittently, which can serve to make the bass level appear inconsistent.

To avoid falling into this trap when you adopt an LCR approach to panning, you just need to resist the temptation to use panning for separation. Check mono playback from time to time, and listen to how the sources panned to the edges relate to each other and to those in the middle. Separation, where desired, might best be achieved using EQ, or perhaps ducking, sample replacement, level automation, mutes or even rewriting some parts. Just not by panning. Attending to such issues in this way won't rob the stereo mix of impact at all, and that sense of width inherent in LCR panning will remain, but it will improve the mono listening experience.

Balancing Act #1

We should also consider the idea of 'balance', in a couple of senses of the word: the balance between the left and right stereo channels, and a change in the overall mix balance that always occurs

When a stereo mix is summed to mono, there's always a change in level of panned signals relative to those in the centre. A lot of people seem to think that this can be corrected by changing the project's stereo pan law — but that actually has no bearing on this balance change!
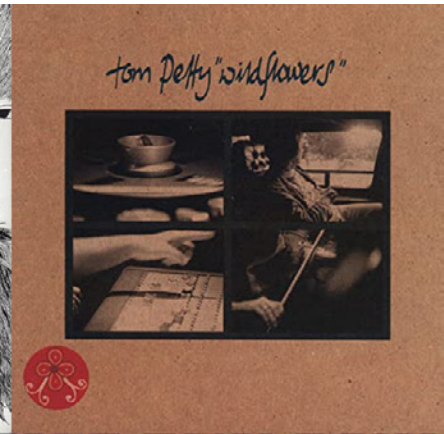
Stereo Sources

It's often pointed out by critics of LCR that any instrument that's been recorded using a stereo mic technique, or the output of a genuinely stereo effect, can't possibly conform to LCR 'rules': unless everything's in the centre then whatever your pan pots are doing the source will appear somewhere between the speakers. Well, that might be true in theory, but it's not a problem in practice. You just need to decide whether these parts should be treated as stereo (pan the channels left and right) or mono (pan both channels to the same point or, depending on the source, ditch one channel). In the former case, where you LCR-pan everything else, there'll be more space in the stereo mix to accommodate these parts. In the latter, you still gain the benefit of fast decision making.

Why not have a listen to some of your favourite records and see if you can pick out some LCR-panned mixes? Here's a rather random selection of very different-sounding but successful records that appear to have been mixed this way, and you might be surprised by how many you find.

» between the various sources, the overall mix balance changes. Specifically, when you switch from stereo to mono, sounds panned to one side or the other become quieter relative to those in the centre. It's a phenomenon that affects every stereo mix, but can be more noticeable in LCR mixes because the harder sounds are panned the greater the difference in level.

Why? Well, the centrally panned sounds are contained equally in both the left and right channels, by definition. So the summing process doesn't simply combine the hard-left and hard-right elements; it adds two lots of the centrally panned sounds too. Consequently, anything in the centre will be doubled in level (+6dB) compared with anything hard-panned to one side or the other. Put another way, anything panned hard-left or hard-right in a stereo mix will measure 6dB lower in the mono sum, relative to sounds panned to the centre. And a 6dB amplitude difference is audible not just to golden-eared professionals but pretty much every listener...

"Can I counter that by using a different stereo pan law?" I hear you ask. No, it's purely a result of the maths of summing a stereo mix to mono: the stereo pan law makes absolutely no difference in this context. The pan law is just about compensating for the change in the perceived level if you move a sound source across the panorama, by introducing a level drop in the centre to compensate for the electrical or acoustical boost when the sound appears in both channels. It's relevant if automating pan positions or using an autopanner plug-in,

but for fixed pan placements you'll inherently compensate for any change in level due to the pan law when you're setting the channel faders.

Central Perk

So what are the implications of this balance change? First notice that I described it earlier as a 'phenomenon', not a 'problem'. There's nothing wrong with the mono mix sounding different from the stereo one. As long as they both work, that's fine, and I would always urge you to focus primarily on making a great stereo mix.

Happily, the change in mix balance will usually work in your favour, because it will slightly emphasise the vocals and any other mission-critical sounds, such as lead instruments, the bass, kick and snare. So everything you need to get the listener tapping their feet, nodding to the beat and singing along should still work well.

That said, there are a few things to listen out for, especially if you're inexperienced at mixing. First, note that stereo reverb or room ambience tends to dry up noticeably in mono, and that can often change the character of a mix quite

significantly. But, given the smaller mix real estate in mono, that can often be helpful.

Another thing I commonly observe is that wide opposition-panned distorted guitars can seem to sound that bit duller in mono. While that's partly due to the drop in level, where it's too much I usually take that as a sign that I've not got their relationship with the bass quite right, or that perhaps I should have had another guitar double in the centre. In this scenario, you might consider a touch of parallel distortion on the centrally panned bass, to inject useful energy in the centre to compensate for the drop in energy of the guitars (you also might revisit your guitar EQ settings in light of that). Or perhaps you could route your guitars to a stereo group bus and use mid-sides EQ to shift some of their energy to the centre, without compromising their sense of width.

You should also pay close attention in mono to the levels of differently panned sounds that have an important relationship. For instance, if you have two fairly exposed acoustic guitar parts playing picked counterpoint, panning them hard-left and hard-right would probably work well: there'd

be no change in their level relative to each other, even though they'd be quieter relative to the lead vocal. Panning one to the centre and the other hard-left, on the other hand, would result in a 6dB change in the level of one part's notes relative to those of the other, and that might (it's not guaranteed to) have an audible effect on any rhythmic interplay between them. Similarly, if you pan multiple percussion or rhythmic parts to different places, you'll want to listen for changes to accents in the overall rhythm.

If you notice any such issues and feel they constitute a genuine problem in a mix when heard in mono, you should start to counter that by changing your panning decisions: opposition-pan those two guitars, rather than placing one in the centre, for example.

Departing From LCR?

Another option I'd like to suggest, which risks offending LCR purists, is to consider straying in a controlled way from your LCR principles as you near the end of the mix, because bringing a hard-panned part closer to the centre will obviously reduce

the part's level drop relative to the centre. I'd urge caution, though: if you were to go through your mix and revise all the parts' pan positions in this manner, there'd have been little point treading the LCR path in the first place, so don't sacrifice the impact of your stereo mix if your mono mix works. Rather, I have in mind that sometimes this is the easiest way to tackle specific balance problems: listen first, and take action only if necessary.

It might be, for example, that panning an offending part to 75 or 50 percent instead of 100 will solve the problem acceptably in mono without reducing the overall impact of the stereo mix. If you have quad-tracked guitars or backing vocals, you might try bringing two doubles closer to the centre while leaving the other pair panned wide, and then experiment with the balance between the two pairs using their faders and maybe EQ. Again, if such moves do the trick and don't have a negative impact in stereo, you're good.

If the mono mix sounds unacceptably dry, then while you could obviously try

adding more reverb in the centre, you might also try narrowing some stereo reverb channels a bit, or panning some mono ones in a touch. You might also experiment with your reverb decisions more generally: maybe move a part to the centre but send it to a wider-panned mono reverb or delay, or to another stereo effect such as a harmoniser. Or vice-versa: leave the mono part out wide, but send it to an ambience reverb patch, and pan the reverb return somewhere in the natural 'pockets' between the LCR panning points.

Summing Up

I love LCR panning in many ways and would encourage everyone to try it at least a couple of times. As I said at the outset, it doesn't always make mixing easier, but I reckon it does encourage you to make better mixing decisions and often to make them more quickly too. Some will find that they like working this way, while others might not — but even if you don't I'm willing to bet that the experience will help you get better at mixing. ▰▰▰

Don't Sweat The Small Stuff!

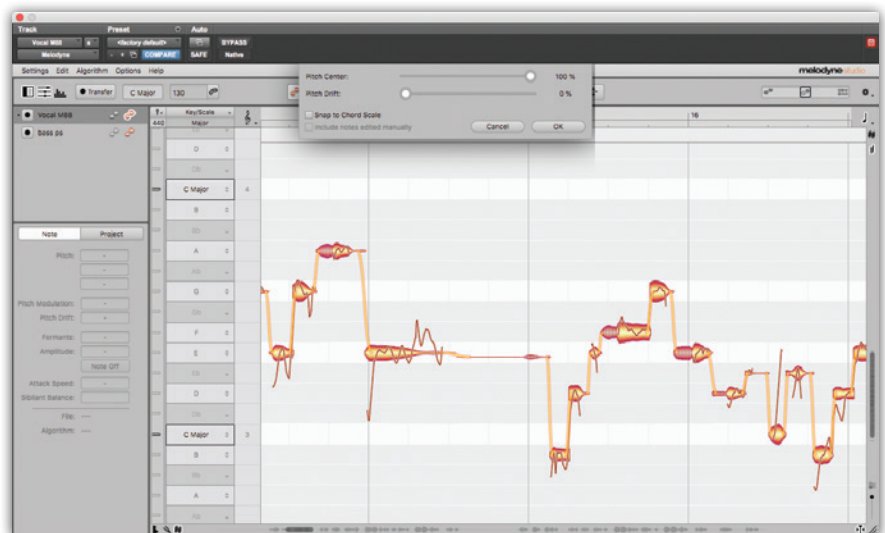## 12 Things People Worry About Too Much When Mixing

MATT HOUGHTON

The only universally applicable rule of mixing is that stimulating an emotional response in the listener trumps all other aims. While it's obviously a great idea to seek tips, tricks and guidance when you're learning how to mix, and to consider how you might apply that advice, you also have to be careful that in the process of applying it you don't lose sight of the big picture. With that thought in mind, here, in no particular order, are 12 things that I reckon many

▬ Pitch-correction processors grow ever more capable, but use them to iron out every vocal imperfection and you risk losing all the character too.

people often worry about a bit too much when they're mixing.

Vocal Tuning: We all now have access to pitch-correction software like Auto-Tune and Melodyne and can, if we wish, make a recorded vocal sound pitch-perfect. But great vocal performances have imperfections, and if you focus too hard on ironing out every one you'll often be left with a part that lacks soul and emotion. What's more, as our ears are particularly well tuned to vocals we notice processing artefacts in them much more than in other sounds. Judging by the reader demos I've heard over the years, many people would be better off paying far more attention to the tuning of instruments, and allow the main event a bit more wiggle room!

Timing Correction: It's a similar story with timing correction. It can be a drag on two fronts: if you spend ages quantising or minutely nudging individual notes to a grid or groove template, only to discover that you've sucked away any sense of groove or human feel, you'll have wasted time and lost perspective. If that sounds like you, try backing off a little: fix what needs fixing, and ignore what doesn't.

Mono Compatibility: Yes, every mix must work when summed to mono and you can glean very useful information
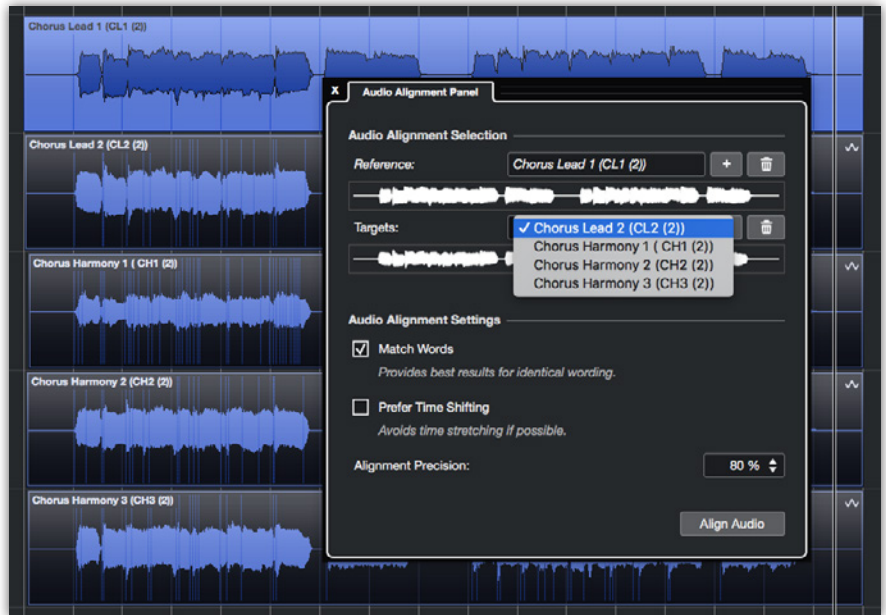
about your mix by checking what it sounds like during mono playback. But in mono your mix needn't sound perfect and it will never sound as impressive as a good stereo mix; it just needs to work. If it sounds great in stereo, and in mono you can still hear the lyrics and the vocal

attitude, feel the groove and pick up on any important melodies, that's arguably good enough.

Nixing Noise: From hiss and buzz to breaths and sibilance, fret noise and finger squeaks to traffic noise and barking ≫

🟥 Sophisticated noise-removal tools such as iZotope RX 8 can be really helpful at times, but focus on using them only when needed or you risk burning time and energy, and losing your focus.

dogs, it's amazing what can grab your ear's attention when you're mixing. And sometimes that razor-like focus on small details can be an asset. But do you really need to spend time obsessing about the removal of such noises? If you can only hear it in solo, forget about it. Keep that focus on the big picture, and you'd be surprised what you can get away with on individual tracks. And even when everything else is playing, consider whether it's really a problem, or if you really need to be distracted by fixing it now.

Using Every Source: If you're given a busy project to mix, it can be tempting to think that you have to use each and every part that you're given: that you have to include all the mic signals, squeeze in every overdub and double, and accommodate each musical idea. While you must obviously respect the artist's creative vision and you'd hardly choose to nix the lead vocal, you don't have to use a part just because it is sitting there in your project. In fact, it's common

for artists to lay down more ideas than a song needs, and if you feel a part doesn't serve the song, the vocal or guitar doesn't need to sound double-tracked, or that extra kick mic channel isn't adding anything, you should feel free to can it.

Hearing Everything All The Time: There's no golden rule that says every part you include in the mix must be heard all of the time — so if you've been fighting to achieve that aim, ask if you really need to spend your time and energy on the 'problem'. Some of my favourite tracks have sounds that weave in and out of each other, sometimes individually identifiable, and sometimes blending to create a completely different sonic

texture. In dense mixes, you can often simply use automation to draw the ear's attention to a part when it comes in, sneakily bring it down again to make space for the next detail, and nudge it up later to remind the listener it's there. You might even find that a lot of parts can be edited very assertively too: does that pad or guitar really need to play right on through every verse and chorus?

Masking: This is the phenomenon whereby when two sounds have content in the same frequency range and play at the same time, the louder sound obscures or 'masks' those parts of the other. The effect is real, and it will sometimes require your attention. But your ears should tell
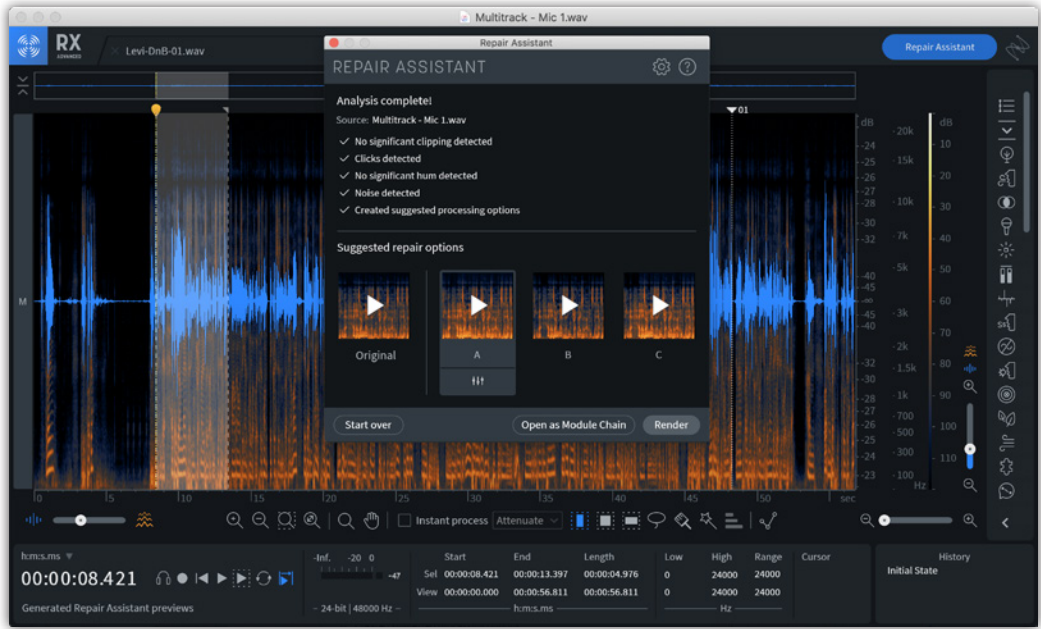
🟥 Just because you have four kick signals it doesn't mean you have to find a space for them all!

🟥 Don't be scared to use your DAW's mute tools to free up some space or introduce some variation.
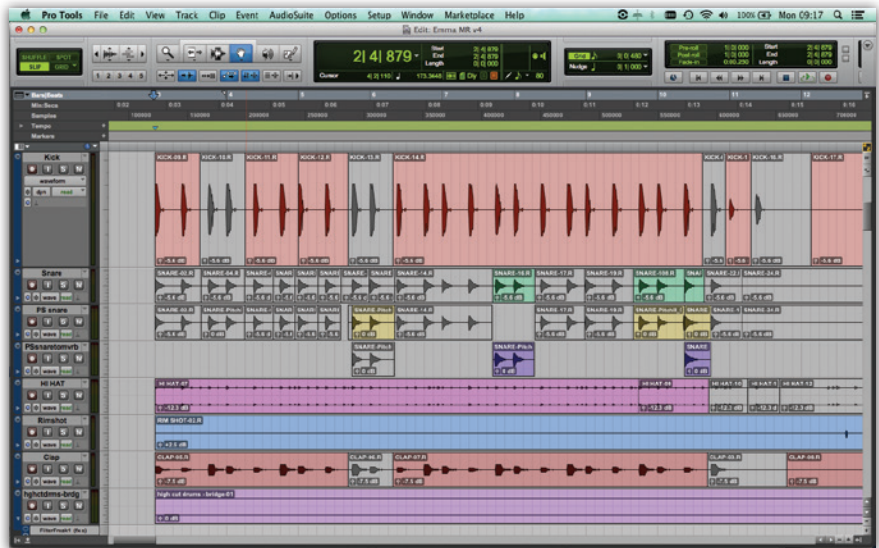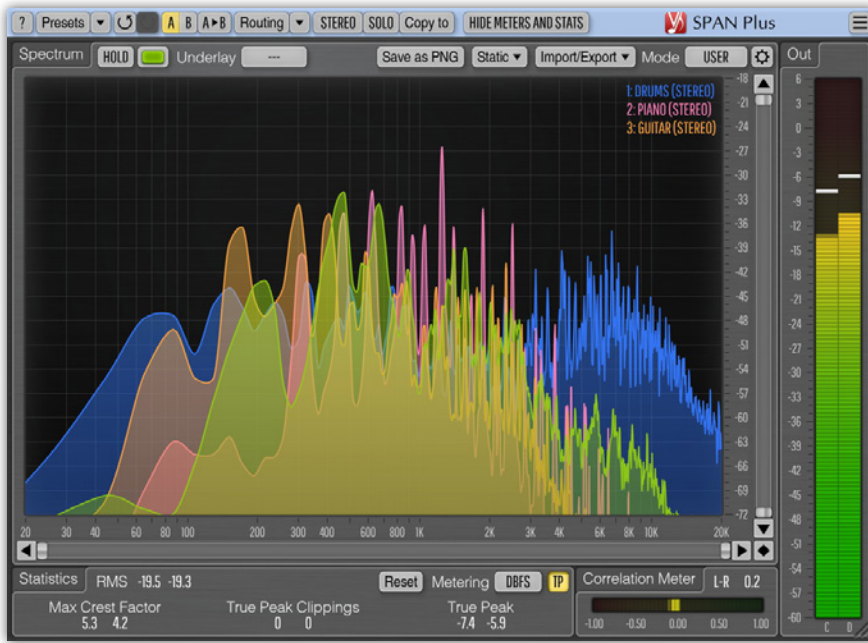
▬ If you hear masking problems, plug-in analysers such as Voxengo's SPAN Plus can help you pinpoint where they are: but try not to go searching for problems you can't hear.

▬ Do you really need to spend time contemplating how to pan the drum kit? Probably not...

》 you when that's the case — you shouldn't need to go hunting in your meters to find such problems. Happily, the most important sounds are usually mixed louder, making many such overlaps less problematic. Where it is problematic, a simple bit of fader riding can often provide a swift solution.

Meters & Analysers: Level meters, stereo correlation meters, frequency analysers... They all have a place, and can be great learning aids. But don't get too hung up on what you see: the bottom line is that what matters is how things sound, not what they look like. If you can't hear a resonance there's no need to pull it down. If something's out of phase, that only matters if the mix *sounds* wrong on mono playback. You don't need to think about loudness while you're mixing, and if your DAW has peak meters on every

▬ Metering plug-ins have a use — but can also be unhelpfully distracting!

channel you can pretty much ignore them as long as you've left a little headroom when setting levels. So yes, learn to interpret what your meters are telling you and feel free to use them when the need arises — but don't let them slow you down or hold you back.

Drum-kit Panning: Should you pan it from the audience perspective or the player's perspective? Do you even know if the drummer's right or left handed? Quite frankly, who cares? Maybe the drummer does, but if it sounds good few others will. In fact, most of the time I find that keeping an acoustic drum kit very narrow in the mix works better anyway

— the contrast tends to make guitars and synths sound wider!

Sweeping For Resonances: A tip I've often seen doing the rounds is to sweep EQ boosts around in search of undesirable resonances and, when you find one, turn that boost into a dip. While this can work, to an extent, it can also waste a lot of time, as you end up attending to problems you can't hear. Worse still, you'll often find that you're not really focusing on the big

Why spend ages sweeping big EQ boosts in search of unwanted resonances? There's a risk that it will lead you to focus on things that aren't causing problems. And if you can hear specific problems, modern EQ plug-ins can often help you pinpoint the problem area much more quickly.

picture while you perform the sweep. If you really need to dip a resonance you can't pinpoint by ear, a good frequency analyser will probably be a much quicker, more accurate way of identifying it.

Translation & Referencing: I see lots of pictures of home studios with several sets of relatively inexpensive monitor speakers, and often different headphones and various 'grot boxes' too. The mind boggles: imagine the quality of a listening

Secondary monitors can certainly play a useful role in your studio — but only if you understand what they're telling you!

system that money might have bought! A secondary monitoring system can certainly be helpful — but only if you know what it's telling you. You need one good system you can trust, and anything else should tell you something specific. For example, can I still hear the kick bass on a small speaker? Or does the mix still work in true mono? There's no point checking a mix on a million different systems and trying to get it to sound perfect on all of them; you'll end up doubting your ears, chasing your tail and losing perspective. The same goes for reference tracks: they should tell you something specific, not be a target to aim for and obsess over. If your mix sounds great and you're tapping your feet, who

cares how closely it matches whichever big-name artist's track you had in mind?

What Other People Think: If you ask people for an opinion on your mix, they'll feel obliged to come up with something to say. That means they won't listen to it in the same way they'd normally listen to music, and their feedback might not reflect their instinctive reaction to the music. Ask five people for opinions and you risk getting five sets of contradictory feedback. Consider feedback, by all means, but make your own judgments: you can't mix by committee! Ultimately, if you're happy and the artist is happy, that's what matters.

JULIAN RODGERS

New features in the 2020.9 update.
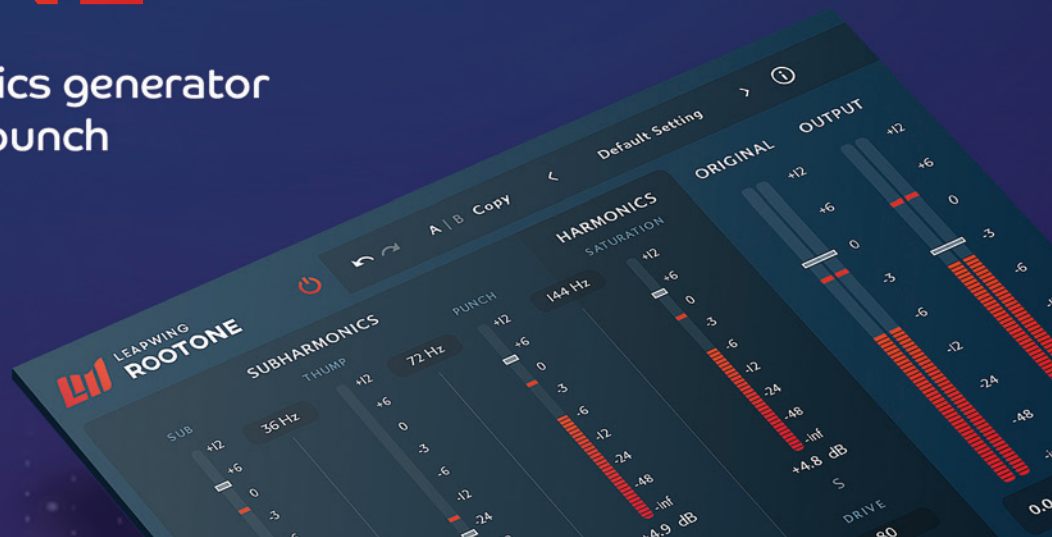
R eleased in September, Pro Tools 2020.9 coincided with a new version of EuCon, and brought proper support for Folder Tracks in the Avid Control app. This, for me, has been the most significant new feature, though version 2020.9 also made some changes to the way Pro Tools works with Cloud services, as well as adding a couple of exciting new features and incorporating the usual bug fixes.

Ableton Link support is the first of these new features and is the easiest to appreciate, as it brings this simple but addictive feature (already in use by Ableton, Reason, Bitwig and many more desktop apps) to the rather longer-in-the tooth Pro Tools. The idea behind Ableton Link is simple: it's a way for devices to share a common time reference across a local network. By 'time' I mean in the sense of tempo; this isn't to do with sample clock. Because the "beat clock" is shared across a network it can be shared wirelessly, and it is in this guise, as a way to keep iOS devices playing together in time, that it is best known, but there are many products, including some hardware, which are Link enabled.

Ableton Link

The performance-based focus of Ableton Link doesn't stop at impromptu iPhone jams, though I can confirm that these are great fun. The possibility to sync VJ applications and lighting equipment, hardware such as the Akai MPCs, as well as more open-ended software like Max and Supercollider, means that Pro Tools can now join this Link-based creative party. Using Ableton Link with the ELK music operating system is very welcome as I've been keeping an eye on this particular technology. If you haven't heard of ELK check it out, the potential is very interesting but beyond the scope of this article.

To set up Ableton Link you first need to show the new Ableton Link controls, which can be displayed in the Transport Bar or in the Toolbar. Simply right-click on the Toolbar or click the drop-down menu on the right of the Transport window.

◾ Clicking the new Link button will enable Ableton Link, which you can use to sync Pro Tools with a number of third-party apps and hardware.

The most important control is the Link button underneath the Link parameter display area. Once Link is enabled and that button is blue, not much will happen until another Link-enabled device or application is detected. In my case this will be the free Launchpad app on my iPhone. Whatever it might be, the procedure is similar in that you have to

> **"The performance-based focus of Ableton Link doesn't stop at impromptu iPhone jams, though I can confirm that these are great fun."**

locate and enable Ableton Link in the settings of the compatible app or device. When that is done the sync bar at the bottom of the Link window will start to cycle from left to right and the number of connected hosts will be displayed, in this case just the one.

If your connected Apps support it Pro Tools can transmit Start/Stop messages to start and stop connected apps remotely. The Quantize value might confuse as it controls the timing interval in which Pro Tools responds (on the beat, on the bar etc), not when connected apps respond. Obvious when you understand what it is, but worth noting all the same.

Although in this case I'm discussing connecting a single Link-enabled app to Pro Tools, Ableton Link has the potential to get very complex indeed. There is no predetermined limit to how many devices can be connected, and I'd imagine that your desire to connect new apps and devices would run out before you ran out of network capacity.

Open Arms

The second new feature introduced is a new behaviour of the record arm

buttons in Pro Tools. To understand the significance of this it's worth talking about the record arm button's old behaviour, and the issues this updated behaviour seeks to solve.

Under the hood, recording MIDI and recording audio are quite different things, and Pro Tools makes this difference apparent in a few places. Retrospective MIDI record is one such example. First introduced in Pro Tools 2018, this allows you to add the MIDI you just played even if you weren't in record. Just hit Shift+C and MIDI data played during playback will be added to your session. Something similar is possible with audio but for that you have to be in one of the punch record modes.

Another way in which recording MIDI and audio differ can be found in the default behaviour of the record arm buttons. Prior to 2020.9, track record arm buttons were latched by default for audio tracks, but with the option in the Operation tab of the Pro Tools Preferences to switch the audio track behaviour to non-latching, where clicking a new record arm button on a different audio track cancels the previous record enable — something referred to by those old enough to remember the old-fashioned mechanical buttons on car radios as 'radio button' behaviour.

The issue was that MIDI and Instrument tracks were unaffected by this preference and always remained in 'radio button' mode. The divide between what happened to audio tracks and what happened to MIDI tracks always remained.   »

◾ Once Link is set up, the green bar will ping from left to right to show that everything is in sync, and the number above the Quantize settings shows you how many Link devices are connected.

Aston Element

"A game-changer. It's the first giant killer of the new decade." - MTM

"A breath of fresh air in the affordable microphone market" - SOS

$199.⁹⁹

FREE shock mount

**FREE** pop shield

10 YEAR WARRANTY

FREE 10 year warranty

◾ In earlier versions of Pro Tools, MIDI and Instrument tracks were unaffected by the Latch Record Enable Buttons option, which made it all too easy to accidentally overwrite a take.

» The irritation felt by some with this is that if you had an audio track record armed and you record armed a MIDI track (or vice versa), the audio track's record arm status wouldn't be automatically cancelled by record arming the MIDI track. Many accidental overwrites have happened this way.

In 2020.9, there is now a drop-down menu in the Preferences window which offers three alternatives: Latch Audio Tracks, Cancel Previous On Audio Tracks and Cancel Previous On All Tracks. The first two options replicate the pre-2020.9 behaviours of latching or non-latching audio tracks, but with the operation of MIDI tracks remaining non-latching and separate from the status of audio tracks. The third option, Cancel Previous On All Tracks, gives the missing option of record arming an audio track cancelling a MIDI track or a MIDI track cancelling an audio track.

Find Your Voice

Although it was quickly resolved there was, on release, an unintended issue with the Standard version of Pro Tools 2020.9, involving voices. This is an area which has caused issues in the recent past as there was an issue in 2019 with the increase in voices for Pro Tools Ultimate and Pro Tools HDX. Whereas the issue in 2019 affected Pro Tools Ultimate users using HDX hardware, this issue affected users of Pro Tools Standard software. Ever since there has been more than one version, the naming of Pro Tools software has had the
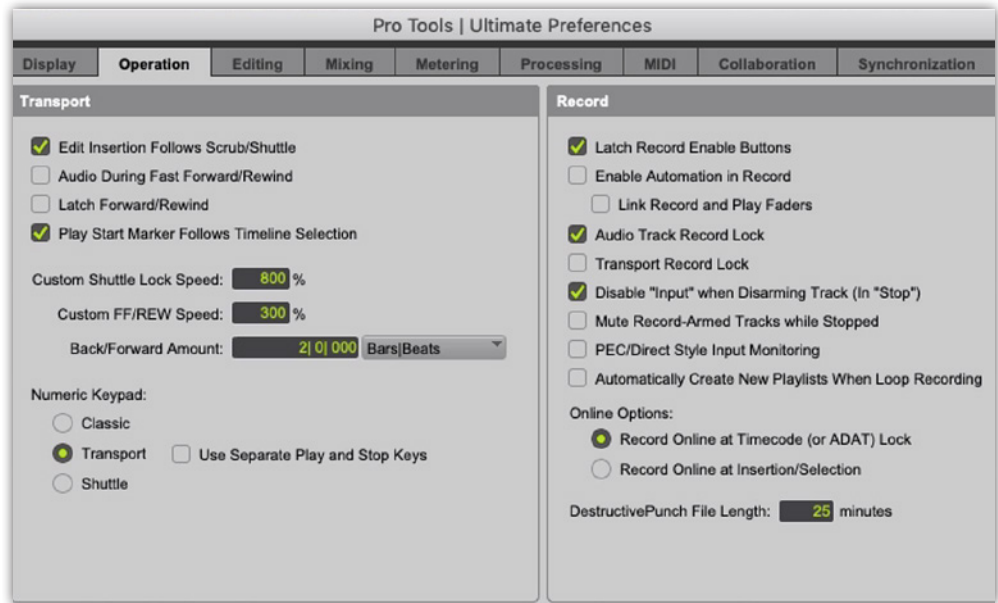
potential to cause confusion so, to be clear, what is being referred to here is Pro Tools, as opposed to Pro Tools Ultimate or Pro Tools First.

The issue was that the voice count of Pro Tools Standard had accidentally been halved from 256 to 128. This halving was unintentional and was addressed quickly by Avid, who released an updated version (2020.9.1) within days, but it is worth being aware that this issue exists and that there is a fix already in place.

Where restrictions exist, most DAWs refer to track counts, but the way Pro Tools deals with tracks is closer to the way that keyboards deal with polyphony, which is understandable considering Pro Tools' heritage as a hardware system. The

situation at present is that Pro Tools Standard 2020.9.1 (the updated release) offers 256 voices, and software-only (ie. non-HDX) Pro Tools Ultimate allows for 384 voices. If you are running stereo tracks this number would be halved, and if you were running at 96kHz the count would be halved again, and then halved again at 192kHz. It's easy to see how the additional voice packs Avid started offering in 2019 for Pro Tools Ultimate software-only users can become necessary. Immersive formats like Dolby Atmos eat through those voices fast! ▰▰▰
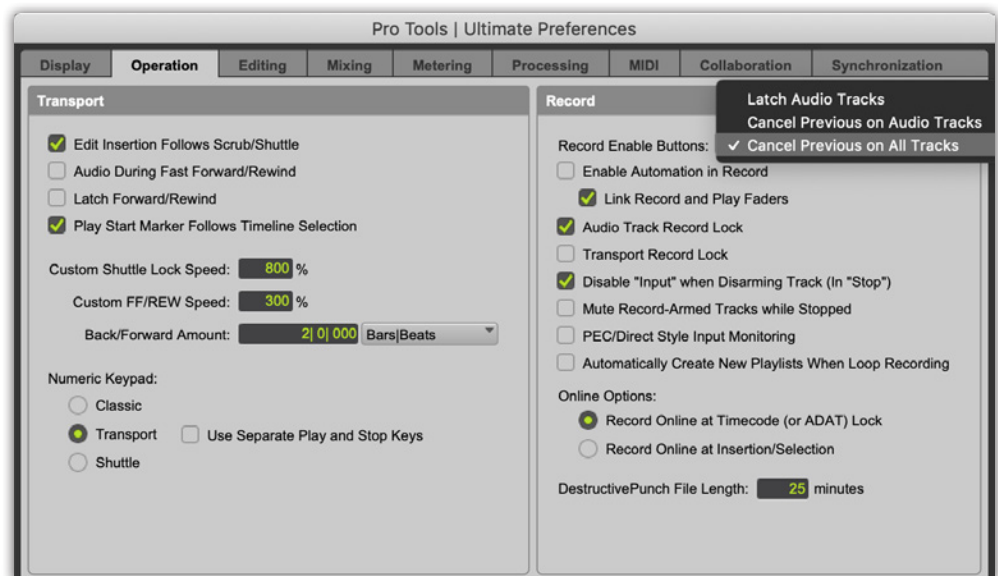
◾ The new Cancel Previous On All Tracks option lets you apply the same record arm latching behaviour to both audio and MIDI/Instrument tracks.

ATH-M20x ATH-M40x ATH-M70x ATH-M50x ATH-M30x
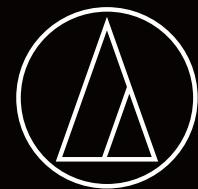
Monitor, Mix, Master

M-Series Professional Monitor Headphones

audio-technica.com

audio-technica

ROBIN VINCENT

One of the brand-new features of Studio One 5 was the Score Editor. Composers and the musically adept could now view their music in a more traditional form. This has been possible in other DAWs, but it was brilliant to see it implemented here, and it looked great and seemed to work well albeit with a somewhat limited set of editing tools. There were a number of things missing but probably the most obvious was the ability to print. This seemed remarkably odd to me, and so I moaned about it in my review of Studio One 5. Now, I'm not saying that PreSonus were so profoundly challenged by my remarks that I spurred them into action, but it's interesting to note that just a couple of months later, version 5.1 drops and it happens to include score printing and a few more things besides.

So, with that omission rectified let's get into the scoring side of Studio One and see if we can produce a page or two of decent looking sheet music.

What's New

The Score Editor has a couple of lovely new buttons. One gives you all your tracks laid out as beautiful pages of music, and another shows you just the part you're working on. It's all labelled up with the name of the project and your name, and it makes you wonder where that information came from. You can change the text by clicking on it, and it will throw up the Song Setup window which contains all the metadata about your song like title, artist or, in our case, composer. I swear I've never seen that window before — which goes to show that you can learn something new every day. Add your details, maybe an inspirational picture, and it's reflected in the score (the text, that is, not the

New features in the Score Editor.
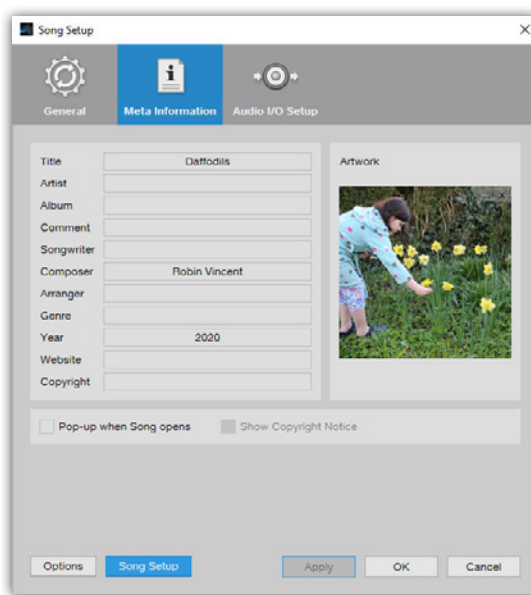
Metadata such as title, artist and composer that you enter into the Song Setup window will be displayed on the score.

picture). You can't move the text around or anything, but it's all sensibly placed, you can change fonts and size, and it looks perfect for a printout.

The third new button is the Print Score button: hooray! You can print out your individual part or the full score simply and easily onto a sheet of paper or to a PDF. Now that our score printing fantasies have been fulfilled let's look a bit closer at what you can do in the Score Editor.

Basic Editing

As I noted in my review, the editing tools are pretty basic and very few action commands have been carried over from the piano roll editor. You can add notes by selecting the length value from the toolbar and dropping it in with the pen tool. You can also edit the pitch of notes very easily by grabbing the appropriate one and moving it up and down. Dealing with time is a different thing. You can't scatter notes about like you can in the piano roll — they have to make sense and be placed correctly or Studio One won't play ball, but that's precisely how it should be when writing a score. Once you tune into the sense that when placing notes on a staff you have to do it properly and in a musical context, then it's rather easy.

Over on the left of the Score Editor we have two new tabs that give us Track (part or staff) information and Layout information. Click in a staff to select it and then you can do notational things like define how the staff is named and what abbreviation is used on subsequent pages. The staff type can be chosen here too, and you can now opt for Grand Staff for a piano track (which has moved from the right-click menu in the previous version).

Lastly, there's Transposition, which allows you to change the key for a transposing instrument. So, for

instance, a brass instrument like a trumpet or trombone will sound a Bb when it plays a written C. As you've probably written the part using a keyboard then it won't be in the right key if someone plays it on the actual instrument. Transposition will change the score to the right key for that instrument and show the correct notes it should play. It does all this without changing the notes being played back through your MIDI sound source or virtual instrument, as this is only for the benefit of the print. You can check back in the piano roll editor and see that the notes are unchanged.

Presets

To make it all easier there's a whole bunch of presets you can apply from a drop-down menu, which cover pretty much every instrument. It'll drop in the staff type, clef, transposition and common name and abbreviations; it makes swapping instruments about and printing new parts a real doddle.

Although there are presets for guitar and other fretted instruments, there's no tablature or option for adding chord symbols. There is an option for a drum or percussion staff and presets associated with it, but the note heads remain as regular notes. To change the note heads to reflect percussion notation you'll have to select the notes and use the Notehead Symbol tool to change them by hand.

The other tab is called Layout, and this acts on the whole score. You can choose between a continuous or paged score, whether to include tempo marks, and whether transposition is shown. If in Page view then you can select the page size, orientation, margin size, note size and spacing, and what font you're using. There's enough here to shape up a convincing-looking sheet of music. You can also enable and disable transposition globally here in case you want to work on the score in the same key and re-enable the transposition before printing.

Multi-measure rests are available when you opt to view a single part by selecting the 'single track view'. These pull together all the consecutive empty bars into a single multi-measure bar with the number of bars it refers to displayed above the line.

Studio One 5.1 introduces a Key Signature track to let you enter and keep on top of key changes and time signatures within your project. This is available below the timeline in the main arranger window along with other utility tracks such as Chord, Marker and Tempo. While the
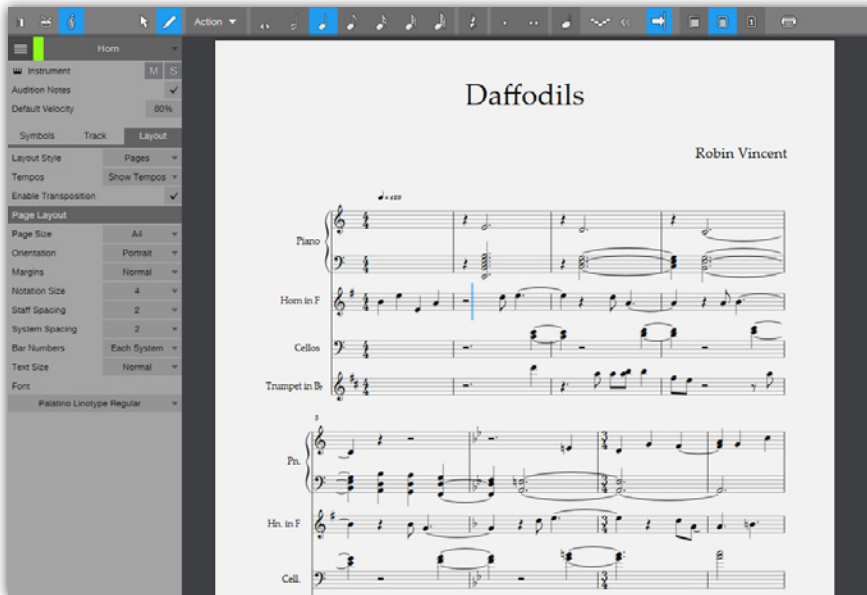
The new score page view and layout options.

Signature track is not present in the Score Editor, any changes you make to the key or time signature are immediately reflected in the score.

Notion Way

Further editing and many more advanced features are to be found in PreSonus' Notion scoring software. There's a dedicated button under the Song menu where you can send selected parts directly to Notion for editing. Notion 6 is available at an additional cost, although if you've opted to become a PreSonus Sphere subscriber then Notion is already included.

Onwards & Upwards

There's enough I think in the 5.1 update to satisfy most people's needs for some rudimentary score editing and printout. Had these features been present in the original release then I wouldn't have had anything to complain about. PreSonus say that there

The Signature track, showing time and key signature changes reflected in the Score Editor.

are more updates to the score to come, which is very intriguing and I wonder how far the integration of Notion will go. For me, I'd like to see the chord track reflected in the score to give guitarists an easier way to follow along. Then perhaps lyrics would be the next step, which, along with the chords, would let us produce a complete song sheet.

There are some new track visibility features that have not made it to the editors which might be useful. The Track Filter can be enormously powerful when dealing with the large track counts you would have when working with an orchestra. You can filter down what tracks are visible by name, selection, loop range and solo, and one particularly useful one which is 'Show tracks with events under cursor'. This means you can put the time cursor anywhere in your project, and you'll only see tracks that have an event at that moment. It would be brilliant if that filtering was reflected in the Score Editor. As it is, in either the piano roll or Score Editor you have the standard track list where you can manually turn tracks on and off, but no filtering. And while we're thinking about filtering, the ability to show only certain orchestra groups would also be useful — so strings or brass instruments, etc. I imagine this is the sort of thing that Notion would sort out for you, though, so if you work with scores a lot, Notion would be well worth considering. ◢◢◢
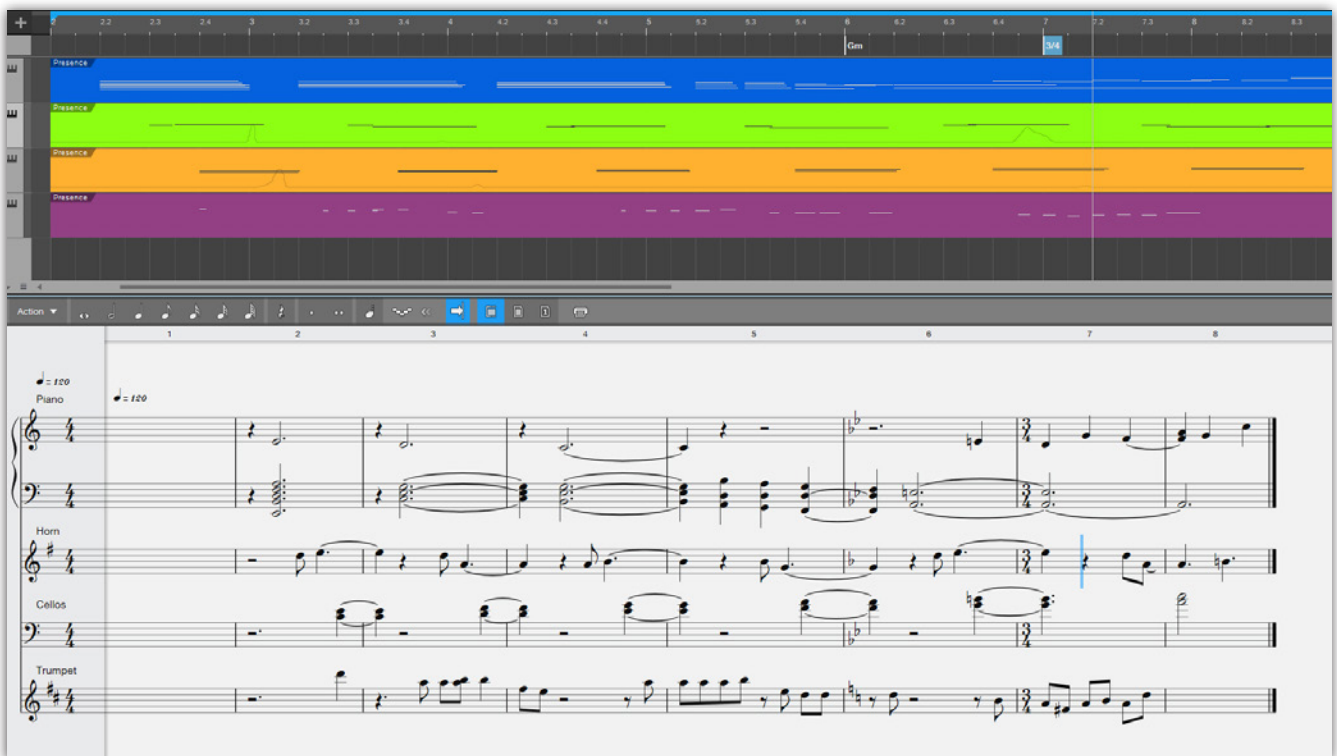
Adding analogue warmth with the Reason Mixer.

SIMON SHERBOURNE

I've been thinking about how many people like to bring a little analogue warmth to their in-the-box mixes, and whether this was something you could do with just Reason's stock toolkit. This is also a good excuse to revisit one of the underused features in the standalone Reason app: the Mixer's Inserts section.

Some DAWs have analogue console or tape emulation built into their mixer, like the HEAT feature in Pro Tools. There are also many plug-ins that add console or tape character, such as the excellent Brainworx_bx series. Some mixers like to use analogue hardware summing mixers on their stems, which itself can be emulated with Waves' NLS plug-in.

Bus V Source

Reason's main mixer is a tribute to a classic large-format SSL console. Its EQ and Dynamics channel processing is based on this. However, the actual signal path through the mixer does not alter the sound character; it doesn't attempt to emulate the colour imparted by the analogue circuitry and amps in the original hardware.

It does, however, have a master compressor, which operates on the final mix bus, and this does provide some of the magic character of the classic 'bus compressor'. This is usually described as 'gluing' the mix together, making all the tracks sound like they are pulling together and adding some overall oomph.

There are plenty of other effects you

1. The Inserts Section of each Reason Mixer channel links to a dedicated device area in the Rack.

could add to your final mix to add some warmth or grunge. Many Reason users routinely use the MClass processors on their final mix, of which the Maximiser has a Soft Clip function. However, if you're shooting for the classic analogue studio sound, the trick is to treat the individual tracks of your mix individually. In the real thing, your channels would all be interacting differently with their own particular console and/or tape path, depending on each source's dynamic characteristics.

Insert Here

Managing effects and plug-ins inside the mixer is handled differently in Reason compared to any other DAW. There's no familiar column of slots for inserting and selecting plug-ins. Plug-ins in fact are completely independent of the mixer. Reason's emulation of a hardware studio extends to the console: there's a single insert point where you can patch in an effects device (or chain of effects) from the Rack.

As we've explored in the past, the conduit between the mixer and the Rack is the Mix Channel device. Every channel in the mixer has a corresponding Mix unit in the Rack. Audio tracks always have a Mix channel and Rack object (with no physical inputs). New instruments get a mixer rack by default (but it's not compulsory), and if you create a mix channel directly in the Mixer it will also have a corresponding Mix Rack device.

The Mix Rack devices represent the audio connections into and out of each mixer channel, and this includes an Insert point. The Mix Rack devices have a 'Show Insert FX' button on both the front and rear views. Clicking this reveals the insert connection points, and a container area

where you can drop the actual effects devices. Incidentally, you can connect cables to the Insert jacks from devices that aren't in the Mix Rack container, but you won't be able to map their controls to the Mix channel encoders (more on that in a moment).

This is a lot of preamble to establish the concept, but it's actually quite simple to use. The main mixer view provides quick access to insert effects from a dedicated Inserts section on each channel (screen 1). This area provides a number of controls but let's start with the Edit Inserts button at the bottom. Clicking this will jump you to that Mix channel in the Rack view and flash the device to make it easy to spot. If there are already insert devices in the channel, they will be immediately available to adjust, much like opening a plug-in in other DAWs. If not, you can drop effects devices into the Insert container and they'll get auto-cabled into the signal path.

Multitrack Tape

Let's get back to our mixing example. Reason has some built-in devices that you can try for some harmonic analogue character or tape saturations. Let's start with Audiomatic (screen 2) using its Tape setting. This device is quite simple: the Transform knob adjusts the tone, and Dry/Wet lets you adjust the amount. We're looking for subtle here, because you're adding this to multiple channels and the effect is cumulative.

Reason doesn't have a quick way to duplicate the same effect to each track. If you copy the device in the Rack the cabling does not get connected automatically, and it would be time consuming to do this manually. However, the Inserts section of the mixer has

2. Audiomatic's Tape mode can be effective if used subtly.

a preset-loading system. Once you've set up your effect on one channel, you can save it as a patch, then load it into all the other tracks in the mixer.

Each channel also has an Inserts Bypass button, so you can quickly hear how the mix is being affected by all this analogueness. Chances are you'll want to make some adjustments. As we've seen, you can jump to the mix effects on any channel by clicking the Edit Inserts button. However, we can do better than this by using the custom controls on Reason's mixer, and make it seem like the effect is built into the console...

Control

The Inserts area of each Mix channel is like a specialised version of Reason's Combinator device. As well as being a container that can hold a chain of effects, it has a Programmer where you can assign four knobs and four buttons to parameters on the enclosed devices. In screen 3, I've opened the Programmer view on one of the Mix channels with my Audiomatic tape effect. After selecting the Audiomatic in the Programmer's device list, I can map the knobs and buttons to Audiomatic's controls.

I've put the Dry/Wet mix amount on Knob 1, the Transform parameter on Knob 2, and mapped the Bypass switch to Button 1. I can set ranges, so by setting Button 1's Min amount to 'Bypass' and Max amount to 'On', I've created an 'In' control for the effect. Avoid mapping the Off position for this parameter, as this stops audio passing altogether.

Again, this setup can be saved as a patch and loaded into all the mixer channels, as

in screen 4. I can now tweak my tape effect quickly across the mix, and experiment without having to keep opening up individual devices.

More Devices

Once you've got the basic idea you can experiment with other devices for adding harmonics, saturation, drive and warmth of various flavours. An old favourite for any of these is Scream. For this application, I'd recommend turning off the Body and EQ Cut sections of Scream, and concentrating on the Tube and Tape modes. You probably want these close to their 'cleanest' settings, which means putting the Tube mode's Contour and Bias to maximum, or the Tape mode's Compression to minimum and Speed to maximum. You can then use the Damage control to set how much grunge is applied.

Pulveriser is Reason's more modern audio degrader. In this context you'll probably want to keep the Filter set to Bypass and the modulation sections inactive. Gentle use of the Dirt effect works rather nicely, and subtly darkens sounds which can be very effective for

getting sounds to sit back into the mix. A little dose of Squash adds some punch.

If you wanted to keep your options open, you could have both the Audiomatic and Pulveriser devices chained in your Inserts, and map the channel strip controls to both of them for easy bypassing and adjustment. This configuration can then be saved as a patch, or even saved into a template song with mixer channels already in place. You could even build a mixer template that emulates a summing mixer approach, using a small number of mix subgroups containing your favourite effects. ■■■

STEPHEN BENNETT

Almost as soon as music began to be recorded digitally, engineers felt they were losing a certain 'something' from not driving the audio through tubes, transistors and transformers. As the emulation of analogue electronics has improved, many have moved completely 'in the box' for convenience and recall. It's perfectly possible to create fine recordings using computer-generated audio alone, as many a number-one hit record will testify, but many of us still feel the loss of that 'mojo' that, we believe, analogue hardware can provide. The cost of adding this hardware to our studios has fallen dramatically in the last few years, with clones of classic gear and the 500-series format providing entry-level hardware that can be bought for very little. So, if you've ever wondered what adding analogue hardware would bring to your recording and mixing, there's never been a better time to find out!

There are several ways you can use hardware with Logic Pro X: recording vocals or instruments, mixing, and mastering. If you treat these as three separate processes, a little hardware, re-used, can go a long way. It's not always obvious in Logic Pro X how to go about incorporating hardware into your workflow and much of the process will depend on your audio interface. Many interfaces have quite complex routing software, so a good peruse of the manual is recommended. If all you want to do is add a little compression or EQ on the way in, a basic 2-in/2-out interface will suffice, but you will need to have an insert point between the preamp and the analog-to-digital converter, or an external preamplifier, to achieve this. If you want to

Incorporating external hardware.

use external hardware when mixing, you'll need more inputs and outputs. If you get hooked, you'll need a lot of inputs and outputs!

If you look at the Input menu in a Logic Pro X Track, you'll see the number of ins and outs that your interface's driver is reporting to the DAW. The first thing to do is give these useful names using the I/O Labels item in the Mix Menu (Screen 1). If you are going to use specific ins and outs for specific hardware, here's the place to make sure you always know where you're sending and receiving your audio.

Mixing

Mixing is where I think the 'mojo' is most obvious when using hardware. I personally find that mixing into a hardware bus compressor, with other devices strapped across various separate tracks in a mix, really helps me find the 'right' sound much more quickly than using plug-ins alone. So, a basic mix session
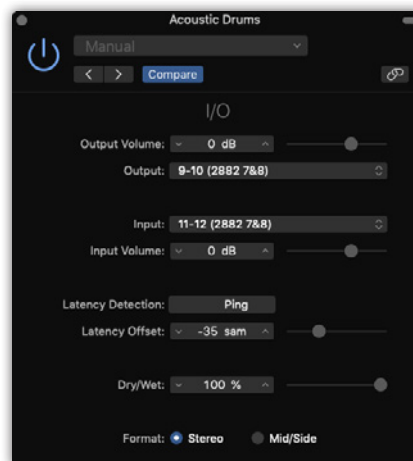
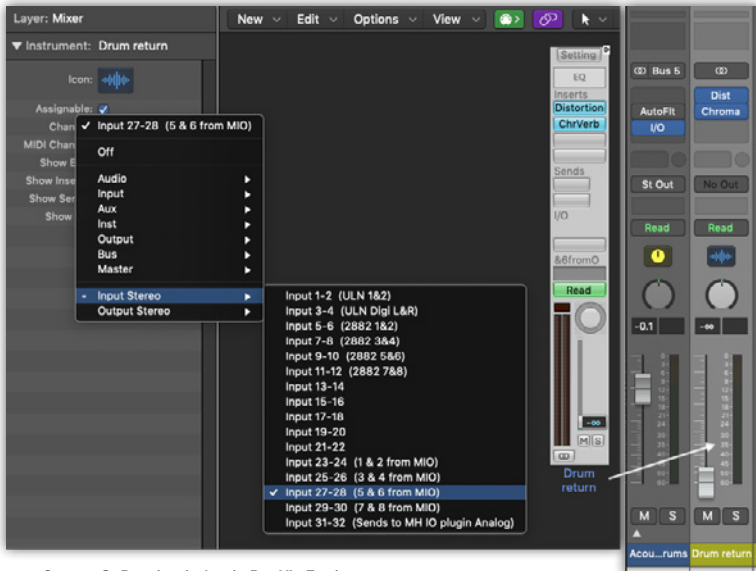

Screen 2: Logic Pro's I/O plug-in.

might have a stereo compressor and EQ across the drum bus, a compressor and EQ on the bass and individual drums, a compressor and EQ on the vocals and another compressor (and perhaps other processing too) on the mix bus. To send audio to and from Logic Pro X to these hardware devices, you'll mostly need to use Logic Pro's I/O plug-in, shown in Screen 2.

If you insert this on, say, a drum bus, you can set the input and outputs of the physical interfaces that you've plugged your bus compressor into. Send and return levels enable you to adjust the amount of gain, which is especially useful when you're trying to hit a transformer input hard for some 'colour', and to adjust the level returned to the track. A nice feature is the Dry/Wet control. If you have a compressor on the vocals, say, and want to try parallel processing, you can adjust the blend between compressed and dry signal here. You will also need to hit the latency detection Ping button before playback begins. This sends an audio 'blip' through your hardware and back to Logic Pro to gauge the time the signal takes to pass through the interface and hardware and back. If you add another processor to the chain you'll need to do a Ping again. Even if the latency detection is a few milliseconds out, it can play havoc with phase.

If you add a plug-in before the send to the I/O plug-in on a track, a thus processed sound will be sent to your external hardware. However, if you want to process the audio after it's travelled through the hardware, you'll need to don a monk's habit and enter that arcane
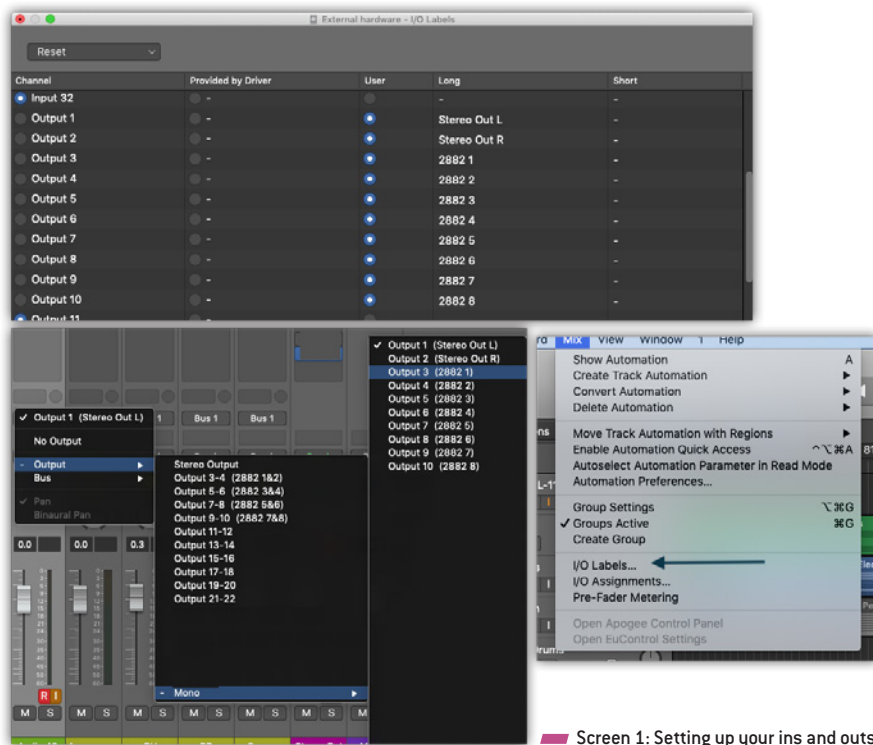
Screen 1: Setting up your ins and outs.

Screen 3: Routing in Logic Pro X's Environment.

place called Logic Pro X's MIDI Environment...

First, send the audio from the track you want to process to the physical output your hardware is connected to, using the I/O plug-in but with its input set to off (the '---' in the drop-down menu). Create an audio track (stereo for a two-channel processor, of course) and give it a useful name. Keep that track selected, and click Window / Open MIDI Environment. Find the Mixer layer and the selected audio track and then, from the Inspector's Channel menu, change the

track type to Input Track, set its inputs to the ones you've plugged your hardware return into, pull down its fader and leave it as 'No output'. Back in the Mixer, you'll now see audio running from your recorded tracks, into your hardware, back through the input track you created and then out through the stereo master, as in Screen 3. Adding plug-ins to this new input track will allow you to process the audio after it has passed through the hardware.

Summing

Now, summing is quite a controversial process, with some saying it's a complete waste of time, and others claiming you need dozens of outputs to make it worthwhile. But with low-cost 500-series

transformer-based summing boxes and inexpensive high-quality mixing desks available new and used, you don't need to spend a lot to see if summing will be right for you. Obviously, if you're using all your inputs and outputs for mixing, you're not going to be able to use these for summing as well. The obvious solution would be to buy extra inputs and outputs (ADAT expanders are one possibility), but there are other ways to get around this issue without spending more moolah.

Let's assume you have eight channels of I/O available and an 8:2 summing device. Once you are happy with your mix, you'll need to bounce down the hardware-processed tracks and busses to audio files by soloing them and using the Edit / Bounce menu. Although Logic Pro has a 'bounce in place' feature, this won't work with hardware, so you'll need to render audio in 'real time'. If you are using hardware across a bus, just solo this and bounce it down. Now re-import the audio back into Logic Pro — I like to record it into a new Project Alternative (which you can create in the File menu). Once you have your bounced tracks as audio, you can then re-patch the physical outputs into your summing device.

You're unlikely to have enough outputs, or summing device inputs, for all of your tracks, so you're also going to have to submix some of your tracks to busses. For example, if you have eight outputs available, you could assign these thus: two to a stereo drum bus, one for vocals and effects, one for basses, two for guitars and two for keyboards. To route tracks to a bus, select them, then select a bus from the Output menu and rename the automatically created bus to something useful.

To send the audio from your busses to the hardware, select the relevant outputs in the bus's Track Output box. To return the mix from your summing device, patch its outputs into two inputs on your interface, load an I/O plug-in on your Stereo Master track and set its input to these physical inputs, and mute the output (Screen 4). You can also insert effects after the I/O plug-in to further process the summed return. If you want to add plug-ins before the return to the Stereo Master, you'll need to create an Input object in the Environment, as before. You can also, of course, add more hardware on the outputs of your summing device for extra processing.

Patch Notes

How you 'drive' the inputs of your summing device, and the specifications of the device itself, is what will ultimately decide for you if summing is worth the extra effort involved. Your wallet is your oyster. For me, using hardware when tracking and mixing is a no-brainer. It makes it easier to judge the mix, albeit at a cost of immediacy and, to an extent, recall. Summing is harder to justify, but if you have a mixer lying around (I use my SSL X-Panda) it's definitely worth a go. But be warned: if you do like what analogue hardware can do to your sound, you might need a bigger rack — and a patchbay!

8 Physical outs patched to 8 inputs of the summing device

8 channels routed to 8 physical outputs

Inputs from summing device

Screen 4: Routing for summing in Logic Pro X.

TalkBack

Jay Watson

WILLIAM STOKES

With albums by band Pond and solo project GUM released in 2019 and 2020 respectively, it's almost impressive that Perth's Jay Watson has any time to spare for his integral role with antipodean psych-titans Tame Impala. The multi-instrumentalist and producer has spent the last few years ping-ponging between London, LA and Perth, developing his skills while cultivating a keen appreciation of what he lovingly dubs 'fun junk'.

At the moment I can't stop listening to
Shiny Joe Ryan, who's my bandmate in Pond: his solo album, which we just got mastered two weeks ago. That album is called *Shiny's Democracy*, which is a fantastic title. As for other music, I can't stop putting on this one Can album when I do the dishes, *Future Days*. I've been listening to it for a decade or so, but I just find it's, like, the ultimate cooking or cleaning album! As you get older and more boring, like I am, that's almost the main backdrop for listening to music now.

There's a heap of stuff on *Future Days* that I can't work out how they did, with the machinery of the time. There's that weird 'wup wup wup' sound at the end of the first track, and part of me thinks it's just a tremolo on an amp and they're playing along to it, but then it sounds so gridded and metronomic… I don't know. There are bits in there that sound like they're from the late '80s, even though it's from the early '70s. It's quite futuristic for the time.

The project I'm most proud of
I think the Pond album *The Weather*. Pond wasn't a joke band, but in our heads, we just liked things like *The Mighty Boosh*,

and every time we wanted to do something 'serious' it didn't seem like it fit the band. We'd slowly worked toward it for years and there's still ridiculous music on it, but I felt like we pushed through and worked really hard on making it into a big, beautiful piece of art. Something to be proud of, in a serious way. Almost every song on the album, there's a bit in there that hits a point, which felt really inspired and meaningful at the time. We've kept that up with stuff we've done since, but that felt like a turning point. We started to sound more like ourselves and less like the things we were ripping off!

The first thing I look for in a studio
It's quite simple, I think. It's just stuff that looks fun to make music with. A studio might have the world's greatest console

> ## Jay Watson: "I'm quite shallow when it comes to recording gear. I think: the cooler it looks, the more I want to plug it in."

or something, or the world's greatest collection of Neumann mics, but if there are no cool drums or synths or guitars… Not that you need that to have a good studio; but you need gear that elicits inspiration, hopefully. In our studios, some of it's expensive and some of it's junk, but it's fun junk. And looking at it, you just want to plug it in and have a go. I'm quite shallow when it comes to recording gear. I think: the cooler it looks, the more I want to plug it in and the more I want to get out of it. That's just me. Music has always been really visual for me. So making it with something you feel cool about, that can go a long way to generating cool ideas.

A friend, Montero, he and I made an album in London when I lived there, in Mark Ronson's studio. He's a friend and just let us use it when he was out of town. And I went in and it was like, 'Holy shit!' It was

like, really fancy stuff, but the coolest fancy stuff. Prophet 5s and big old RCA ribbon mics… Just the best and coolest stuff from each decade of recording, really. And I feel like in half of that album, when I listen to it now, I can hear the glee in the playing. All the toys!

The biggest misconception about the role of the producer
The biggest misconception is that it just means one thing. It's only recently that I've become comfortable with writing 'produced by' on records. Some records are produced by people who did almost nothing, and there are other records where the producer has co-written all the songs and played most of the instruments. I think that's the thing I find frustrating: people can assume that if I've produced someone's record I've done more than I've actually done — or, if someone else has produced our record, that they've done more than they have actually done. It's too wishy-washy! I like 'engineer'. Or, if you've helped write or helped arrange it, then you get credit for that.

My go-to reference track
I can't think about recording bass without thinking of the Beatles' bass sounds. It's so annoying to mention the Beatles, but I feel like they hit a bass sound and that's just it. From Revolver all the way to the end of the Beatles, it's there. You need the 'donk'. Whenever I set up my bass in Pond, I have this preamp on top of my bass cab that I can dial in the EQ on, so I'll dial in the 'donk' and play 'Dig A Pony' and if it sounds right then it's good to go!

My desert island studio item
It changes all the time, but most recently I bought a real Roland 808 drum machine.

I wasn't trying to buy one, I bought it because it was a really great price, and I thought to myself: it's a lot of money but it's an investment! And it's in really good condition. I'm in love with it. I'm amazed at how versatile it is. You can use it to do that really tinny, like, Suicide drum machine sound, and you can make the most stonking house and hip-hop beats on it. And it never sounds daggy. Do you guys say 'daggy' in England? It's kind of like… socks and sandals. Dorky. Anyway, it never sounds daggy on anything! It's way more versatile on any type of music than I thought it would be.

The piece of gear I'd bring back into production

I can't think of one specific piece of gear, but I do often think a lot of equipment is made too well now. It doesn't fuck up in the way that a lot of the old bits of gear I have do. Like, an old mixer; they distort so easily, and that's a big part of the sound and what makes stuff you run through it sound really badass. And the same with old synthesizers: I've got this one called an MG-1 Moog Realistic. It's a really old,

cheap, simple synth, but it sounds like a synth that someone has built themselves in the shed at the back of their house, you know? I would kill to have one of those where you could save presets and we could use it live in Pond, and where it didn't go out of tune. But it's always this trade-off with something super modern and reliable that just doesn't sound fucked-up enough.

I think if I had my own audio company it would aim to make gear that functions perfectly well but sounds sort of broken. I like performances and mixes that sound like they're going to fall apart at any second, and I like equipment that sounds like it's going to fall apart at any second. It's like when the battery is dying in the fuzz pedal, and everyone gets all excited and you try and get the take down before it actually dies. If you could somehow get that with the mod cons of new equipment, then I would like that.

The producer I'd most like to work with

I have a lot of respect for J Dilla. I think a lot of the music that's come since him, whether it's Kanye or Jai Paul, or Caribou

— stuff that I really love — you can just hear how a lot of it comes from that J Dilla stuff. I think, as highly rated as he is, I think he's still weirdly underrated. He's like the Jimi Hendrix of hip-hop production.

The advice I'd give the me of 10 years ago

Not to be so concerned about what you've already done, or to dwell on the mistakes you've made. From an engineering standpoint, the biggest thing I'd tell myself would be: just get yourself a good interface and maybe one really good preamp, and then you can just cruise around with one guitar lead and an XLR lead and it'll sound good. The last couple of GUM records, it's all this one 500-series BAE Neve clone. I also have this pair of these Australian preamps made by Sebatron, which I really like. The guys in King Gizzard & The Lizard Wizard got me onto them. That BAE thing, it was less than a grand and I didn't have anything else, it was just in a little lunchbox on its own. But the records after that sound noticeably better than when I just plugged straight into an M Box. ▰▰▰

NEW YEAR, NEW STUDIO!

21 Ways To Improve Your Studio In 2021

THE SOS TEAM

With the world still busy fighting the coronavirus, professionals and hobbyists alike have been spending more time in our home studios. As well as making more music, we've had plenty of time to reflect on what's working well for us, and what's not. So the *SOS* editorial team shared notes to come up with 21 ways you can greet 2021 with a better studio.

1. Improve your acoustics

Acoustic treatment is an investment that will benefit both your mixes and your raw tracks. Several manufacturers offer full room kits that are a great way to get in the right ballpark if you have the cash, including GIK, Auralex, LA Sound Panels, Acoustics First, Iso Acoustics and Vicoustic. But as a minimum, basic 'mirror point' absorbers are a must, and if you have space, proper bass trapping — not just foam — in the corners will tighten things up.

Most home studios are one-room affairs, and if you don't have a separate live room, you'll need to balance accurate monitoring against a good-sounding tracking environment. Especially in a small space, this usually means keeping the room fairly dry. Use software reverbs to add depth to your tracks, or buy some cheap reflective material that can be set up temporarily when tracking. Even if you can't make permanent changes to your room, there is still worthwhile action you can take. Buy a couple of heavy duvets and hang them away from a wall to deaden the space temporarily, or get your hands on a portable vocal screen such as the sE Reflexion Filter.

2. Upgrade your monitoring

The world of small-studio monitoring has changed a lot over the last decade, and if you haven't kept up with the pace of development, you could be missing out. You may even be working only with headphones, in which case the addition of even a small set of active monitors will give you an invaluable new perspective. Modern entry-level monitors from manufacturers such as Kali Audio, ADAM, Mackie and KRK offer very impressive

Models like Kali Audio's LP8, above, have made quality monitoring more affordable than ever.

performance at a modest cost.

Further up the scale, innovative driver and tweeter materials, a renewed interest in coaxial speakers, and new technologies such as built-in DSP are redefining what's possible in a compact monitor. Check out, for instance, *SOS* reviews of products like Genelec's The Ones, Jones-Scanlon Baby Reds, Focal Trio6, Dynaudio Core 7 and Barefoot Footprint 01.

The bottom end is super-important in most modern music styles. But if you try to get huge bass extension from nearfield monitors, you'll compromise their operation throughout the audible range. So, if you're confident that your room acoustics will handle true bass, you may well find that the best option is to pair your existing speakers with a subwoofer. By filtering out the low bass from your satellite speakers you allow them to shine

at what they do best, and a good sub will let you hear what's going on right down to the fundamentals.

3. Get a secondary monitor system

Even if you already have a capable set of studio monitors, a good secondary speaker or set of speakers can often give you a useful perspective. Maybe it's a mono, sealed-box, single-driver speaker like the newly reissued Auratone or the Avantone MixCube, which tells you instantly what the balance is like in the all-important midrange. And whereas once the car radio or a 'grotbox' was an essential aid to mix translation, the challenge now is to make your material sound good on modern 'smart speakers'. Your audience is listening on Amazon Alexas, Apple Homepods, Google Home speakers and Sonos devices. Do you have something similar to check what they're hearing?

4. Always be ready to record, with a patchbay or mixer

Fighting your way through reams of tangled cables on your desktop in order to plug something in is a real buzzkill when you just want to sit down and make music. A patchbay can really help to keep things tidy, and give you almost instant access to everything in your studio. If you want to know just what the humble patchbay could do for you, check out Hugh Robjohns' article in *SOS* December 2020. Basic models from the likes of ART are super-affordable, while if you want the ultimate in flexibility, take a look at recallable, remote-controlled patching systems from Flock Audio and CB Electronics.

Another option for the small studio is a compact analogue mixer with enough inputs to leave everything permanently plugged in. If you have a lot of synths, a rackmounted line-level mixer such as Tascam's LM-8ST is a particularly compact and handy option, while SSL's neat SiX will bring a touch of luxury to any desktop.

5. Go portable

Sometimes software gets in the way of the creative process. A standalone audio »

Improve your studio by freeing yourself to record outside the studio. Products like Sound Devices' Mix Pre range offer studio-quality recording in a portable device.

» recorder can be really liberating when you just want to get some ideas down. And if it's lightweight and portable, like the Zoom L8 reviewed in this issue, it also means you can capture ideas in high quality anywhere in your house, without lugging around your whole system. Many also double up as mixers and have onboard effects for cue mixes, while others such as the Sound Devices Mix Pre range offer technical quality as good as any regular interface. So you can either use one to replace your interface, or just as another option.

We live in a golden age for analogue synthesis. If you're feeling flush, treat yourself to a flagship instrument like the Moog One and Arturia Polybrute!

6. Add another mic

Many home recordists have only one high-quality studio microphone. If that's the case, is that limiting the quality and variety of your recordings? A large-diaphragm capacitor mic can be great for vocals, but other sources might benefit from a moving-coil dynamic, small-diaphragm capacitor mic or a ribbon mic. If you only have a cardioid mic, perhaps consider an omni one, or something that offers multiple patterns. And if you record instruments as well as vocals, think about the possibilities for stereo recording that open up when you have two mics.

The classic vintage large-diaphragm sound is more affordable than ever thanks to models such as Neumann's TLM102 and to manufacturers such as Sontronics, Avantone, Telefunken USA, United Studio Technologies and Mojave, but the last few years have

also seen many innovative alternatives reach the market. Townsend Labs' Sphere and Antelope's Edge implement advanced mic modelling and Lauten Audio's LS-208 and LS-308 offer unique directional responses, while companies like Earthworks and Josephson Audio continue to push the technological boundaries. Audix's SCX25A fits into

places where other large-diaphragm mics wouldn't, and Sennheiser's Ambeo and Rode's NTSF1 have made high-quality Ambisonic recording available at a new price level. Even the humble moving-coil dynamic has been transformed in recent years, through innovations such as the active circuitry in Aston's Stealth and Element, and Shure's revolutionary Dualdyne capsule design.

7. Get hands-on with a hardware synth

Just because you can use software instruments to create almost any sound doesn't mean you have to do it that way. There's something uniquely satisfying about a hardware synth with hands-on controls, and they're available to suit every taste, space and budget. From desktop instruments like Roland's Boutique series or Modal's Skulpt and Craft, right up to desirable polysynths like the Moog One, Korg Prologue, Nord Wave 2, Sequential Pro V and Arturia Polybrute, there's nothing like a real hardware synth to bring a smile to your face.

8. Looking good, feeling good

We're often so focused on the gear and the sound that we forget that a recording studio is supposed to be a creative,

Enjoy classic-style mics like United's FET 47 (left) or embrace innovations like mic modelling (Townsend Labs' Sphere), affordable Ambisonics (Rode's NT-SF1) and novel polar patterns (Lauten Audio's LS-208).

» inspiring place. A comfortable chair (and another for your collaborator!), good lighting, a lick of paint, inspirational artwork, better storage solutions… all these things can really help to make your space promote creativity and productivity. Carl Tatz Design even make a chair specially designed for studio use!

9. Percussion

In-the-box grooves can be good, but the temptation to loop things or just use what you have available can often lead to things sounding a little flat. Real percussion — shakers, hand drums, tambourines and so on — can be a great way to inject a little human feel, and keeping a box or drawer full of such things needn't be expensive. Just bang and shake into the mic, and layer it over your existing beats where the song needs a lift. Or if you're a real skinflint, use finger clicks and hand claps!

10. Consider correction

No matter how much you spend on monitors and acoustic treatment there will always be limits as to what you can achieve, especially in a domestic space. Likewise, no pair of headphones is perfectly linear out of the box. So why not use one of the many DSP solutions to optimise your monitoring system? Some, like Sonarworks Reference, Dirac Live For Studio and IK Multimedia's ARC, are software-based solutions designed to be used as plug-ins or standalone applications on the master outputs of your interface. Others, such

▬ Soundtoys are one of many plug-in manufacturers who offer tempting bundles across their entire range.

as Trinnov's impressive ST2 Pro, are hardware solutions that are independent of your Mac or PC.

11. Get out of the box

If you're stuck in a rut composing with a guitar, or with DAW software and a MIDI keyboard, try an alternative approach. Grid-based devices and hardware sequencers can be huge fun and are guaranteed to get you thinking in different ways. There are all-in-one hardware options such as Akai's MPC range, Elektron's Digitakt and Novation's Circuit 2, which are perfect for go-anywhere creativity. Hybrid systems like NI's Maschine harness the power of your computer behind a hands-on, grid-based controller, and software such as Ableton Live and Bitwig Studio is made to be paired with a grid-based controller.

12. Try some new plug-ins

Still using only the plug-ins that came with your DAW? There's a whole world of amazing third-party processors, and nearly everything is available to try without risk, courtesy of time-limited demo versions. Novel corrective tools such as Oeksound's Soothe 2, Sonnox's Oxford Drum Gate, SoundTheory's Gullfoss and Leapwing's CenterOne can make almost any audio recording easier to mix and master; or, if you're struggling to learn how to mix, products like iZotope's Neutron and Toontrack's EZmix can give you a welcome helping hand. Manufacturers such as Valhalla, FabFilter, SoundToys, McDSP and Eventide offer attractive deals that will give you an entire library of excellent plug-ins at an affordable price. Others, such as Plugin Alliance and Slate Digital, offer subscription models that give you access to everything in their catalogue »
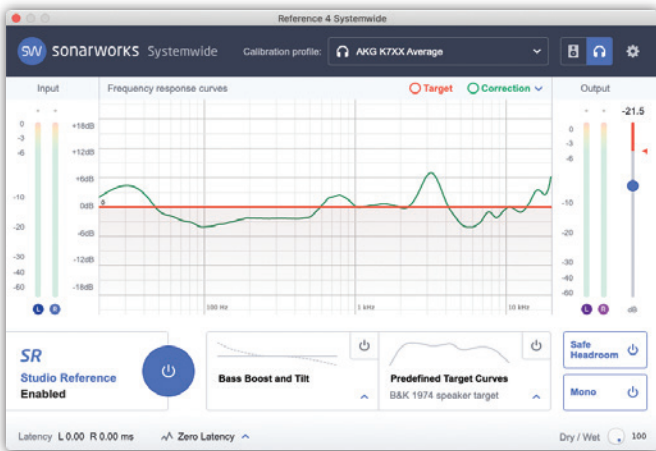
▬ Speaker and room correction can be done using a hardware system such as Trinnov's ST-2, or in software such as Sonarworks Reference.

▬ Classic and modern: BAE are one of several companies making excellent preamps inspired by the Neve 1073 (above), while Audient's ASP800 (top) allows you to record super-clean or dial in as much colour as you like.

» for an affordable monthly fee. And if you want the ultimate in plug-in power, invest in a hardware co-processor such as Universal Audio's Satellite or McDSP's remarkable APB system.

13. Explore online collaboration
Covid means we can't get together in studios and rehearsal spaces so easily. But new tools are making it super-easy to make music online. Enjoy real-time music-making with friends thanks to generic tools like Zoom and Google Meets, or the dedicated Jamulus app. Share DAW sessions in the cloud using Avid's Cloud Collaboration or a service such as Dropbox. And turn your studio into a virtual control room to direct sessions taking place elsewhere with applications like Steinberg's VST Connect and Source Elements' Source Connect.

14. Invest in your input chain
You might have a high-quality microphone, but does the rest of your input path do it justice? If you're currently plugging directly into an interface, maybe you should consider getting a high-quality outboard mic preamp. A state-of-the-art modern preamp such as Millennia's HV-3C will give you clean gain for days, ensuring you get the purest possible capture of quiet sources. Alternatively, you can get some vintage mojo into your tracks by opting for a classic Neve or API design. These are still available from the original manufacturers, and have also inspired affordable recreations by Warm Audio and Golden Age as well as high-quality homages by the likes of BAE, Aurora Audio, Great River and Vintech. Universal Audio's preamps and input channels likewise draw on a classic heritage, while ART's Pro MPA MkII offers valve warmth on a budget. Look for preamps with an output gain control or fader if you want to explore the possibilities of driving the input hard to add pleasing coloration. Cranborne's EC2, SSL's Alpha VHD-Pre and Audient's ASP800 offer the best of both worlds, with the option to introduce colour to an otherwise clean signal.

Finally, don't neglect the less obvious aspects of your input chain. An expensive mic deserves and needs a professional-quality stand such as are made by K&M, and you might be surprised at the difference a good-quality popshield such as Stedman's ProScreen range can make.

15. Invest in yourself!
If lockdown means you have more time on your hands, why not use it productively to improve your own skills? From learning an instrument to ear training to advanced mixing technique, there are resources available online to fill any gap in your expertise. The renowned Berklee College have been offering online programmes since 2002, and offer a huge range of courses covering every conceivable aspect of audio production. Mix With The Masters have recruited an incredible array of A-list names to put together their programme of videos, webinars and seminars, while the Art & Science of Sound Recording course was devised by none other than legendary producer Alan Parsons, and events that usually take place 'face to face', such as Music Expo's Summit seminars, have now moved online. *SOS*, meanwhile, have teamed up with Nonlinear Educators of Canada to provide our comprehensive *SOS* Tutorials series.

16. Fix those dodgy cables
You know — the ones that kinda work and kinda don't! Is there anything more

▬ One fader, many possibilities: PreSonus's compact Faderport can make a serious difference to your mixing workflow.

frustrating than plugging in and hearing intermittent crackles and changes in level? There's no sense skimping on these sort of basics, so either grab yourself a soldering iron and learn how to fix them, or order a few spares and move on. Good-quality cables from the likes of Whirlwind and Hosa cost very little in the scheme of things, but are worth their weight in gold.

17. Fade up your mixing skills
Work faster and more effectively by pairing your computer and DAW with a control surface. There are serious efficiency gains to be made by adding something like PreSonus's Faderport or Frontier Designs' Alphatrack, which give you a single fader for writing automation to key tracks, or a cheap tablet running Avid's free Avid Control app. Softube's Console 1 ecosystem offers more than just DAW control, replicating not only the experience but also the sound of working on a classic large-format console, while systems such as the Slate Raven and Quantica Modula make full use of the latest touchscreen technology.

18. Integrate your guitar pedals into your mix setup
With manufacturers such as Electro-Harmonix and Strymon constantly introducing new and outlandish processors, guitar pedals can be an inexpensive and creative source of effects at mixdown, and it's often straightforward to integrate them with a few tweaks to your setup. They all accept standard quarter-inch jacks, so the main challenge is to present the signal to

EastWest's Composer Cloud gives you access to a huge range of orchestral libraries for a modest monthly feel.

them at a suitable level and impedance. A passive direct injection box such as Whirlwind's IMP2 or Radial's Pro DI used 'backwards' may well be all you need, but there are also dedicated interfaces such as Audio Simplified's Pedal Pusher.

19. Upgrade your headphones
If people are listening to your music, they're mostly doing it on headphones. The boom in headphone sales has led to plenty of new developments in headphone technology, from the 'high-excursion' drivers used in Austrian Audio's Hi-X models to the wholly new and very impressive HEDDphones. Established manufacturers such as AKG, Beyer, Fostex, Sony and Sennheiser continue to improve traditional moving-coil dynamic designs, while new planar magnetic headphones from Audeze and now Avantone offer

a compelling alternative. And there's strong competition from newer names in the field such as Neumann and Focal, as well as Audix's excellent A-series models. If you haven't shopped for headphones for a while, you might be surprised at how good modern cans can sound.

20. Take control of what you hear
Many audio interfaces have basic monitor control built in, but few offer all the features of a mixing console's master section. You might be surprised at how much of a difference a proper monitor controller can make to your workflow. Calibrate your monitors to work at a known SPL, enjoy instant speaker switching, and take advantage of features such as mono summing, left/right mutes and talkback.

21. Hire an orchestra (virtually)
What better way to expand your musical horizons than to bring in a battalion of virtuoso string, brass and woodwind players? Few of us will get to hear our material played by a real orchestra, but sampling technology has come a long way since those ubiquitous 'piano + strings' patches in home keyboards. There is now a huge range of virtual instruments that place the world's best orchestral players at your beck and call. Take a look at Dave Stewart's comprehensive round-up back in 2015 to get an idea of what's on offer, then bring the choices up to date by browsing the catalogues of manufacturers like Spitfire Audio, EastWest, VSL, Heavyocity, Orchestral Tools, Sonokinetic, NI, Project SAM and more. ◢◢◢

New headphone technologies continue to emerge, including HEDD's novel use of the Air Motion Transformer (left) and Austrian Audio's 'high excursion' drivers.

YOU ME AT SIX
Recording *Suckapunch*

For their latest album, You Me At Six threw out the rulebook and spent six weeks at a residential studio in Thailand. Producer Dan Austin was at the helm.

JOE MATERA

Hailing from Surrey, England, five-piece outfit You Me At Six have spent the past decade and more pushing boundaries both musically and sonically. Never a band to rest on their laurels, their ambitious sonic experimentation has seen them successfully fuse a cocktail of dance, R&B and hip-hop elements to a pop-rock template. And they're not afraid to employ whatever means at their disposal to achieve their forward-thinking musical vision. With six albums already under their belt, their seventh and newest album *Suckapunch* saw the band up the ante in all departments, in the process creating one of the most original and fresh-sounding albums of recent times.

"Trying to not repeat ourselves and trying to use different sounds across the songs was a big challenge," explains producer Dan Austin, who helmed the production desk for the album. "If we'd already done something like use a particular drum sound or a particular drum technique then it would be, 'OK what can we do differently on the next song that we're going to record?' while at the same time keeping everything sounding as consistent as possible as an album, as a whole. And there was a lot of programming going on with this album. One of the things we did was to make our own drum samples to program with. And I used a lot of resampling of sounds to make textures and synth beds and things like that. It was time-consuming but it made things sound different between songs and made it sound individual to the record as well."

"It's a bit of a creative freedom on this record, where we can be experimental and showcase what we can do," adds You Me At Six guitarist Max Helyer. "Since we've now been together for quite a while as a band, »

You Me At Six producer Dan Austin.

Photos: Ed Whitfield

Band and producer at Karma Studios, Bang Saray, Thailand.

Guitarist Max Helyer spent considerable time experimenting with his guitar tone, making use of plug-ins in addition to his usual pedalboard.

we felt we had established something for ourselves on the previous records that we're really proud off. But how do we keep it interesting and exciting moving forward? For example, on 'Beautiful Way', some of the inspiration from that song was very much from drum & bass music and electronic music, inspired by bands from Pendulum to the Prodigy and even some old-school rock music such as the angst of Marilyn Manson's 'The Fight Song'."

Due East

Both producer and band chose to record the album at Karma Sound, a residential recording studio in Bang Saray, Thailand, during the course of a six-week period in October and November of 2019. "We looked at various options around the world," says Austin. "I always had the idea that if I was with a band, I always wanted to be in a residential studio. I like everyone being in the same place at the same time and without distraction, where you'd be able to work as late as you want and be able to roll out of bed in the morning and get into the studio. And not have people coming in or anyone going off to go get food at different times and stuff, which can become really distracting for the sessions."

As well as fulfilling the role of producer and directing the sessions with his hands-on work ethic, Austin was and is very much the sixth member of the band. "I had worked on the band's previous album *VI* [*2018*], but with this album they really wanted to tear up what they were doing and try different approaches to recording," he tells *SOS*. "So I became more than just the producer of the band. I got really involved in every stage of the songs. We are always trying to push things outside of what we would have traditionally done before. Trying to not repeat ourselves has become a big thing. And we all get along really well together as people; that's been a big part of why we've been able to achieve what we've been able to achieve on this album."

The band took a radical approach with the recording sessions, creating unique drum loops which lay the foundations for their songwriting. "I have a studio in my house and because I'm really into dance music, techno and house and stuff like that, I will build a lot of tracks from like keyboards, chopped up loops and things like that," explains drummer Dan Flint. "A lot of things we demo start off that way. So what we then do is create our own loops, where I don't play any instruments or any sort of drum tracks from start to finish. I'll only play kick drums, or snare drums, or kick and snare drums together, or just do hi-hats, and create all these loops from just messing around on the top of the tracks. Then I'll cut bits out and Dan Austin and I will sit there for hours, just going through it all to find really cool parts and then piece them all together. It gets really creative and is so much fun, it is my favourite thing about being in a band, and being in the studio with Dan.

"Once we have the loops, I will also need to figure out how to replicate it in order to do it live, as well as build a lot of tracks for the live show. I have to perform them the best way possible, so that they come across the same live as they do on the studio loops. But doing it this way makes it so creative and so unique. If we can build from the ground up, something that is already interesting from just a drum beat, then anything else that comes after it will be also interesting and different."

Plug-in Baby

In the hands of You Me At Six, the digital realm presents options that the band use primarily as creative and inventive tools rather than enhancements. Their approach is evident on the album's opening track, the riff-driven, groove-oriented, 'Nice To Me', which sets
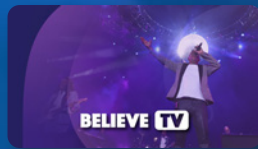
NAMM BELIEVE IN MUSIC
WEEK — JANUARY 2021

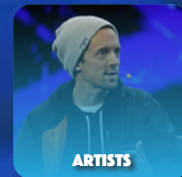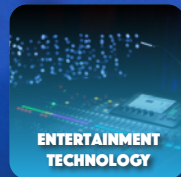## The global gathering to unify and support the people who bring music to the world

▶ BELIEVE IN MUSIC TV

Watch 80+ hours of programming to educate and inspire you

GEAR TV • BELIEVE TV • ARTIST TV

💡 TRAINING

Explore 100+ sessions to help you navigate the 'now' and prepare for the 'next'

OPENING SESSIONS • BUSINESS • AUDIO TECHNOLOGY • ENTERTAINMENT TECHNOLOGY • ARTISTS • ADVOCACY / NONPROFIT / STUDENTS

🌐 MARKETPLACE

Discover the latest gear and connect with the brands you love, including:

Agencia Argentina de Inversiones y Comercio Internacional • AKG by HARMAN • AVID • beyerdynamic • Blackstar AMPLIFICATION • BOSS • Boutique Guitar SHOWCASE • Breedlove • CASIO • CLOUD • Cole Clark Guitars • Córdoba • Crown by HARMAN • d&b audiotechnik • D'Addario • D'Angelico NEW YORK • Darkglass Electronics • Idean Guitars • EBS • EV Electro-Voice • Epiphone • Ernie Ball FOR EVERY STAGE • Event Safety Alliance • ESP • FISHMAN • FOCUSRITE GROUP • Gibson • Hal•Leonard • HARMAN A SAMSUNG COMPANY • Hosa • IK Multimedia. Musicians First. • JBL PROFESSIONAL • KALA BRAND MUSIC CO. • kmc music • KORG • KRAMER • KRK SYSTEMS • L-Acoustics • LYON & HEALY CORPORATION • Martin & Co. EST. 1833 • MACKIE. • Marshall • Nu:Tekt • OWC • PIGTRONIX • REMO • Reverb • Roland • SAKAE OSAKA HERITAGE • SEQUENZ • SHOWMAKERS • SHURE • SONY • Soundcraft by HARMAN • Spector • SPITFIRE AUDIO • Supro • Takamine SINCE 1962 • Tanglewood • TASCAM • Taylor • TRABUCCHI THE VIOLIN MAKER • v-moda • VOX • waldorf • YAMAHA • ZOOM

THE CELEBRATION BEGINS JANUARY 18, 2021.

Global Livestream January 20 | **Marketplace** January 21 - 22 | **Believe in Music TV** January 21 - 22

REGISTER FOR FREE AT
attend.believeinmusic.tv

Suckapunch was tracked through Karma Studios' SSL 4056 E Series console.

the mood for the rest of the album and showcases what technology can bring to the recording process. "On 'Nice To Me', I was conscious of trying to push the sound of how guitars are 'meant' to sound," explains Helyer, "and from that track on, I think we've delivered certain things on this record. Surprisingly this song was actually one of the last songs written for the album before we went to Thailand, and I did it from my home studio. Having gone more into music production over the past five years of my life, it's allowed me the ability to sit there on my computer and mess about with things. You get so used to having an amplifier and pedalboard set up in front of you, where you can twist and turn knobs easily and dial in sounds straight away. But I've really enjoyed sitting in front of a computer, loading in different plug-ins and trying stuff that you wouldn't normally try in order to be a bit more adventurous.

"For example, I've got a really cool DI sound, and this involved me jumping in and trying lots of different plug-ins. I tried to find different sounds that I don't have access to from my pedalboard. So for textures and layers and creating a mood, it was really great to have this setup in front of me where there was no time constraint. When I write a song I will look at it as soundscaping, where I am trying to create a vibe. When I was playing the opening riff on that song, I was thinking in cinematic terms and how I could create a mood. I looked at some of my favourite artists and how they created sounds. I went back and watched films and studied how they created a mood and atmosphere with how they played and the sounds and effects they were using."

Slowing Down

Another element in the You Me At Six toolbox was the use of the varispeed.

Bassist Matt Barnes in the Karma Studios control room.

Suckapunch

You Me At Six's seventh studio album, *Suckapunch*, is released on January 15th via Underdog Records/AWAL.

For Josh Franceschi's vocals, Dan Austin hired a Telefunken U47.

"I used a lot of varispeed on this record, which I really enjoyed as I hadn't been able to do it for a while," affirms Austin. "The studio was running Digidesign 192 A-D/D-A converters with a Digidesign Sync I/O. A lot of people don't have the Sync I/O anymore, and it enabled me to change the speed of the work clock. A lot of the drum takes we recorded were about two tones down. Dan [*Flint*] would play the drums and we'd record them slower and then take them back up to speed. This made the drum tracks sound as if they were sampled. We also did that with some of the guitar tracks as well."

"We recorded some tracks at variable speed, you know that old-school thing that they used to have in all the studios and which nowadays you don't find

»

Dan Austin made extensive use of the studio's microphone collection. Here, Dan Flint's drum kit is being recorded by a pair of AKG C414s, a Shure SM57 and SM7b, a Yamaha Subkick, a Coles 4038 and a number of Sennheiser MD421s.

» anymore there" concurs Flint. "That was challenging in the way that you have to work out how to get a unique sound out of it. But with any of these challenges we have, they're always creative. It is not something we stress out about, we just go with it. A lot of the challenges happen naturally, like spit-balling in the studio. We don't really come across many challenges in terms of difficulty, as we have such a good relationship with Dan that it feels like we're the same person or like he's the sixth member of our band. And it flows so well that it is non-stop creativity."

The Gear

Tracking for the album was done using an SSL 4056 E Series desk, which the studio had recapped just before band and producer arrived to begin the sessions. "The guy who recapped it used non-standard capacitors which gave the desk sound a real chunk to it," explains Austin. "It was a very unique-sounding SSL, so essentially it was modified, but it sounded incredible.

In terms of outboard gear, there were [*Empirical Labs*] Distressors, UREI 1176s... standard fare really. A lot of stuff was done digitally with plug-ins. I normally record fairly flat and then manipulate stuff. I sort of mix as I go along too and the band kind of play into mixed sounds as we go along, so that everyone knows where they're going to end up. Obviously it affects your performance playing or singing into a sound that reacts in different ways. There's a huge amount of distortion on the record too, and a lot of that is the Soundtoys Devil-Loc and Digidesign Lo-Fi. There was a Manley Pultec clone too which had come from George Michael's personal studio, which we nicknamed, 'The George'. I used it as an insert on a mono room mic and I blew it up on the desk. I had the mic pre on full blast and because of the capacitors, it distorted in a particularly pleasant way. Also there was lots of desk EQ added to this signal and then 'The George' on the insert enabled me to sculpt the distortion in a really lovely way. I used this distorted mic on loads of sounds and even on

some vocals as well. So there was this big fat distorted room mic that I could blend into or against my other microphones on any given source."

For Austin, having a selection of mics on offer was integral to capturing the tones he was seeking. "We rented a Telefunken U47 which I took with me because that was the only thing they didn't have at the studio and I needed it," says Austin. "I've always used one on Josh [*Franceschi*]'s voice since we started working together. His vocal chain was a U47, through a Neve 1073 and a Distressor and then into whatever plug-ins I am using in the computer that I'm recording him through. Other stuff was fairly standard, the band has some Telefunken M81's which I absolutely love and use in places where I would have normally use an SM57 historically, like for top and bottom snare and on guitar cabs. I love them and they sound fantastic. The studio had AKG C414s, Neumann KM 84s, Neumann U87s for, like, pianos and overheads and room mics. For the guitar cabs, we used

a Telefunken M81, a Neumann U87 and a AKG 414. Also on the record quite a lot of the guitars were just DI'ed straight into mic preamps with a Distressor over it. And though that makes the sounds quite cold and digital, we were aiming specifically on some sounds to be digital-sounding."

Helyer also brought along his amp of choice to the studio with him. "I've been using The Little Chopper from a company called Audio Kitchen, which is a boutique amp company in the UK," explains Helyer. "And because the amp is so small that I can fly anywhere around the world with it, I took it with me to play through in Thailand. The studio had a Fender Twin Reverb which is also my main other amp that I use in my live setup. So my main setup for the recording was the Audio Kitchen and Fender Twin. I'd normally run them both at the same time. The Little Chopper gave me the low end, the body and warmth and bassiness of the sound and the Twin Reverb gave me the high-end treble cut-through. So the combination of both amps gave me what I wanted. And then how I stack my pedalboard up by arrangement of pedals on the floor, also gave me a different clarity from it too."

Mixing As You Go

Never a band to follow any kind of traditional route, they also employed a 'mix as you go' approach, with the tracks all very much close to mixed by the time the band left the studio. "Either for myself or for any other guys that do the final mix, the method is more about tidying up frequencies and getting it so that it's consistent," Austin tells *SOS*. "In terms of creative decisions, they would've all been made by then. With mixing as I go, there are no surprises, or they're shouldn't be when you get a mix back, especially with You Me At Six stuff, because those decisions have already been made. And all the processing is printed as well. I give the mixers the options of stuff without it, but 99 percent of the time they'd be using the processed version of what I send them.

"And that comes from the fact that, for us, it's not like, 'let's throw down some guitars or some vocals and then down the line we will fix stuff. Because it is a creative process while we're in the studio, it has to be semi-mixed whilst going along. We're creating sounds. We're creating vibes that have to be looked at in detail whilst doing it. It's never an afterthought. Everything has its specific place and if it has to be there, it has to have a real reason for being there."

This approach is integral to the band's ethos of always evolving. "We've been guilty in the past of making records where we've kept the same guitar sound for each song, kept the same drum sound and the drum kit didn't really change," says Flint. "With this way of recording and how we do it with Dan Austin, everything is set up ahead of time, everything's ready to go whenever we want. We might be halfway through some guitar track and we'll be like, 'Well actually, what about this sort of thing here?' We will jump from one thing to another in order to try and enhance the song. It's never, 'OK the guitars are ready to go, let's do the guitars.' Sometimes you will end up repeating yourself so much when you do that. Whereas we really go into details. 'What are we trying to create here? What is the vibe of this song? If this song is alive, what does it want to say and how does it want to say it?' We like to look at each song as unique and give it the best moment in the sun it can have to be as great as it can be."

WILL STOKES

How does a composer with an open disregard for the fetishism of synthesizers go about creating an album that is as dependent on its hypnotic electronics as it is on classical instrumentation? For London-born, Scotland-raised Anna Meredith MBE, the musical success of Mercury-nominated second album *FIBS* was never going to hinge on a specific choice of sequencer or the hunting down of any 'must-have' legacy synth. "I am a bodger," Meredith warns me with a laugh as we, along with guitarist and producer Jack Ross, sit in the quad of London's Somerset House, which also happens to be the location of her studio. "I'm not a compositional bodger, but I am a bodger with some of this stuff!"

The idea of a 'second album' is something of a misrepresentation of Meredith's remarkable career trajectory. Many will doubtless consider 2016's *Varmints* something of a breakthrough, yet the composer has been garnering the respect of both sides of the so-called classical and popular divide for the best part of two decades; far longer than such a slim catalogue of original studio albums might suggest. From 2012's body-percussion piece *HandsFree*, which saw performances at the BBC Proms, Hamburg's Elbphilharmonie and beyond, to high-profile soundtracks for such shows as Netflix's dark comedy starring Paul Rudd, *Living With Yourself*, it's almost remarkable she has managed to find the time to work on any self-commissioned studio records at all.

"I was doing a few little residencies," Meredith explains. "Really focused writing of little 'nubbins;' little cells, germs of ideas. Not too fleshed out, but things I thought had the potential to become stuff. I brought the whole band down, five of us, to the Attenborough Centre for the Creative Arts, and we listened to a few things and chatted about these ideas."

It wasn't long, however, before a succession of impossible-to-ignore commissions became incumbent on Meredith and demanded that the album be temporarily shelved. Projects like 'Five Telegrams', commissioned for the BBC Proms (complete with projection mapping onto the exterior of the Albert Hall), the release of *Anno* ("Half my music and half Vivaldi's Four Seasons," the composer says, "kind of half electronics and half string orchestra"), and the soundtrack for Bo Burnham's coming-of-age comedy *Eighth Grade* demanded the lion's share of the composer's 2018. "So, I put the album to one side," she explains, "and then it was just

ANNA MEREDITH

Composing & Producing *FIBS*

Few artists straddle the line between classical and electronic music with as much success as Anna Meredith. We talk to her about the creation of her recent Mercury-nominated album.

like, clear the decks. Stop touring, let's make some kind of schedule to make it happen."

London Calling

From August 2018 to January 2019 Meredith retreated to her London studio to begin work on the album, which stemmed from an almost completely abstract place. Testament to her exhaustive composition process, at least half the overall time devoted to creating *FIBS* was spent without recording a note.

"I don't think: 'I'll make 30 tracks and pick the best 11,'" She explains. "It'll be: 'I'm going to make 11 tracks; and they're going to be fast instrumentals and slow instrumentals, and there'll be a vocal for me and vocal for the drummer, and there'll be a group vocal...' I know the things I want in it, so then I'm trying to work out the pie chart — the balance of energies — of the album as a whole, so they all balance up. I keep that in check before I've written a note."

Indeed, there is a key stage of Meredith's process that takes place before any writing occurs at all: drawing. 'I do these graphic drawings of the shape of the track, the energy of it. I do this for every kind of music I write. They almost look like rockets! It's like, a non-musical way to get the pacing right. Initially they're almost like mood boards. So, I'll have an A3 bit of manuscript with a little squiggly shape and a few adjectives, and it'll have a few bits of notated rhythm. It might say something like 'stabby' drawn into a shape, or it might say 'big bass here?' and then I might notate underneath a little example of what that might be — a few chords or something, or a rhythm. So, it becomes almost like a schematic. I never think about genre. It's never about a 'techno track' or a 'rock track'. I only think about the compositional ingredients. Something that builds up in a particular way."

The Score

The route Meredith's ideas take from compositional 'nubbins' to sounds in a DAW is an unconventional one by today's standards of studio recording, yet nonetheless highly musical and even traditional. "It's always in Sibelius," she tells me. "It's always notated. Years ago, when I started writing straight into Ableton, everything just became layers. There was no direction. So I thought, 'OK, what is my skill set?' I know how to notate music, I know how to control bars and time signatures and pace stuff well, so I made a decision to go to electronics right at the last minute. It's not designed for playback at all, Sibelius. It does it, but it uses general MIDI, so it sounds awful! But I kind of like that, because it means that if the ingredients are good, then I know that when I upgrade it with real instruments, it'll be a lot better. It's a way to test whether or not things works compositionally. So, when I go into Ableton the hard work is done. I'm not relying on synths to provide the character. I'm relying on composition.

"In my studio... there's like, no gear in it! It's a little MIDI keyboard, a computer, a desk, some manuscript and a piano. It's not filled with synths or drums machines — none of the material things. In fact, the opposite. I'm a little bit daunted by some of that stuff and not particularly interested, it doesn't help me write good music. I used to be quite panicked and felt that was what I was meant to be doing. Almost, 'You're writing electronic music so you should be getting really into this stuff,' which is why it's been great for someone like Jack [*Ross*] to be on it. I'm good with big brush strokes, but not good with tiny brush strokes; so I kind of need someone to help me do the stuff I'm not very good at, and... slightly struggle to care about, which is sort of embarrassing!"

Instrumentation

When it comes to listing her synthesizers, Meredith's vocabulary is refreshingly off-kilter and creative. As if barely concerned with the names of things beyond their musical value, she recalls: "I've got a lot of '90s ravey synths that I really like: analogue, kind of chiptune-y. Because a lot of the music is very fast, I can't have too many wet sounds. So, I'm drawn to a lot of beepy, short, percussive, vintage-y sounds. I've got a lot of packs and plug-ins of vintage sounds and old computer game sounds, lots of the Ableton ones, some Roland stuff, some Super Nintendo-style ones, some Kontakt stuff... Nothing cool! I guess I just don't
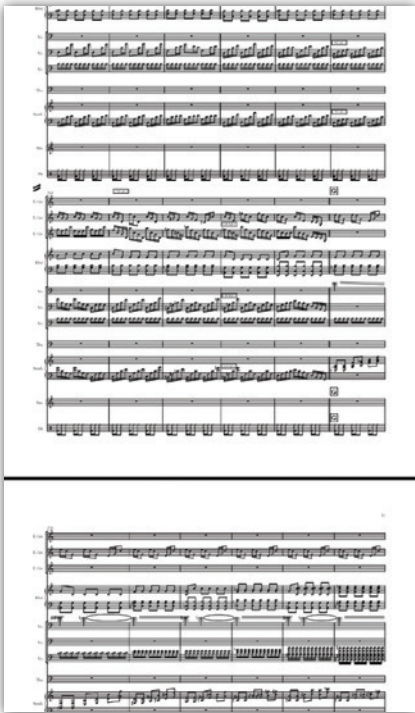
◼ Anna Meredith's compositions have an unusual genesis: they all start out as diagrams, illustrating their eventual structure, before being scored in Sibelius.

let the software daunt me too much, because it then overwhelms the music.

"I think, having a classical background, when I'm writing for an orchestra, I know every instrument. I know the range of what they can play, but I'm not going to think 'What can a violin do that's never been done before?' I'm thinking 'I'm writing for a violin.' So, I think about the mix of electronics in a similar way. It's like my old friend... I start with a preset! I'll move it around to make it my own thing, but it's very simple: 'fizzy synth A,' or 'big fat squelchy synth B.' I'll have these ones I keep coming back to because I like them. It's about finding a functional sound and using things I already know, which function well."

Jack Of All Trades

Guitarist and production aficionado Jack Ross has been working with Anna for a decade. As the composer considers her unconventional approach to electronic instruments he interjects, with equal parts gravitas and encouragement: "Anna, you're different to some people who get buried in technology, but you are no slouch with Ableton! You know how to move around in there and get things that you like. But it's your way round, which is a good distinction to make."

The term 'production aficionado' may seem somewhat hackneyed, but as Meredith points out, "we've struggled with the word 'producer,' because it seems to be a very confusing word. I did all the electronics — nobody else wrote the electronics or had any input on them — but Jack produced the recording. We've really struggled, to be honest, to find the right language. In the classical world, if you're called the producer of a string quartet album, in that sense you'd be the only producer because nobody's composing anything. It's all about recording pre-existing material. But, if you're working in lots of different kinds of commercial music... it feels like it could be many things."

Recording

The recording proper started modestly, in Meredith's own space. "Tuba, cello, guitar, clarinet, all the voices, some pitched wine glasses... basically it was all done one at a time, in a tiny room, layering up, editing," she tells me. "But it was great for me, because I find it hard to listen to anything beyond the composition. When we're listening back to takes, and Jack is trying to talk about which one is slightly better here or there, I'm just thinking, 'What a great line! Smashed that!' So it's so important to have someone who can listen in that way.'"

Laughing, Ross remembers one instance where choosing a take was particularly easy. "We'd done one take of pitched wine glasses, and then went in to have a listen. We shut the door to the vocal booth that all the glasses were in,

◼ Guitarist and producer Jack Ross records a guitar for *FIBS*.

🟥 Percussionist Sam Wilson recording at Somerset House.
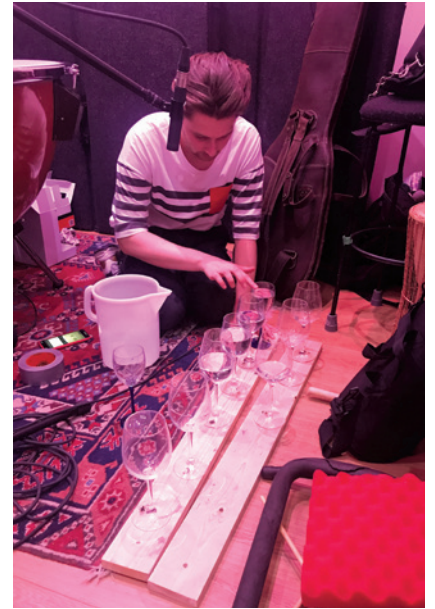
and the primary glass just fell over and smashed. It was like, 'Yep, guess we got it!'

"You get in the zone of listening to EQ or the character of the microphone... and I love being in that rabbit hole. It's quite an addictive sensation!" says Ross. "The studio facility at Somerset House is fairly rudimentary. It's got a tiny space, almost a vocal booth, really, for separation; a pair of Yamaha HS7s and a pair of Bowers & Wilkins 705 S2s; a Midas M32R desk; a handful of microphones... Mic-wise it was, like, three SM57s and a couple of EV RE20s. Also a really nice valve mic with a particular character, an sE Z5600a II. It's pretty bright but with plenty of body, and we used it for all vocals. We used a DPA 4099 on cello and clarinet, because it's a great little thing that you can get really close; very detailed and perfect if you don't have a lot of room to play with. And then I stuck the sE mic in front of both instruments as an option, a different character. We also had an Aston Spirit. Great mic, great company. They're really versatile. We used that on guitar cabs, tuba and cello at different times, depending on what character we wanted to capture.

"I've always been really interested in studios and the recording process," he reflects. "I agreed to record all of the acoustic stuff except for drums. I think you really need to do a good job of that. You need a studio with someone who really knows what they're doing

in that specific space." Enter decorated engineer Shuta Shinoda, at East London's Hackney Road Studios. "I've done loads of stuff with him over the years," says Ross. "That guy, in that room... He really knows how to record drums and make them sound fantastic. He's got beautiful

mics and preamps that he knows just how to work. Lovely old SSL [*4000E*] desk, big [*ATC 100a SLM*] speakers... we had a great time recording in there. In that situation, my role was to keep track of take sheets, give Sam [*Wilson, drummer*] feedback about what he was

»

For the acoustic instruments, Jack Ross chose to use a DPA 4099, usually supplemented with a large-diaphragm mic.

» doing, think about sounds, talk to Shuta about mics and mic placement."

Carefully nurturing the project over that hard-to-define threshold, from individual composition to instrumentalist collaboration, Meredith had prepared well with her players for this stage of the recording process. "I would have a few days with each person," she says, "so, with Sam, we would have written

the drum parts together, figuring it out. It was the same with Jack and the guitar, Maddie [*Cutter*] on cello and Tom [*Kelly*] on the tuba. Figuring it out, just in a studio somewhere, away from recording, and then by the recording stage things are quite fixed. There isn't too much improvising."

"It's about getting good takes," adds Ross. "Solid takes that are going to work well with the electronics. It is an album that is driven by the electronic content that Anna has generated. But all of the

instrumental parts need to find their place and need to support that, and you've got to find where they, kind of, blossom out of that electronic texture and take the spotlight, and also where they just sit back to help the trajectory of it. That was the challenge, really."

Anna considers: 'We're quite internal. We definitely keep stuff in-house." Turning towards Jack once more, she continues, "I was able to talk to you guys more than with *Varmints*. When I was writing, I was thinking, 'Sam's got a great voice, so he'll sing a track,' and then you [*Jack*], me and Sam can sing a track because I know our voices work together... And, I'm also thinking of what Maddie the cellist can do..."

I suggest to the pair that this approach almost harks back to a bygone era of composition; where composers would be more likely to work with the same players time and again, as opposed to the common practice today of working with by-the-hour freelancers. "We all love working in that way," responds Ross. "There are players who wouldn't even consider it, they'd just be like, 'Get back to me when you've got a playable part!' So, that back and forth relationship is really important. Tracking this stuff, we know what we're aiming for. We really know what we want to achieve with it. We know, 'That bassline, that's got to be a tuba, leading that!' So we've then got to find a way to get it to punch through, or to fill a gap in this or that synth sound so that it suddenly feels huge. You lose all of the extraneous stuff, that sense of trying to feel your way through things. It

Anna Meredith (left) with Marta Salogni, who mixed the album.

Marta Salogni: "No ego, just skills."

and processing. She's just a really great person to work with."

Needless to say, Salogni was the right engineer for the job. A resourceful approach and respect for the finite elements before her made for an effortlessly contiguous mixing stage, demonstrative of the crew's maxim when it came to making *FIBS*. When it came to mastering, it was down to John Webber at AIR studios; perhaps best known for his work on a vast array of iconic jazz remasters but also, aptly, responsible for finishing the epic soundtracks of films like *Alien: Covenant* and *High-Rise*.

Looking back on *FIBS*'s recording and mixing processes, Ross reflects: "It was a lesson. Use the stuff you've got really well. Know what you're trying to achieve rather than getting a new plug-in that you think might solve your problems. It's the same as with Anna; she's in pursuit of the impact of the sound and the sonic experience, rather than using gear and plug-ins to inspire something that might be something. You want to be thinking: 'I know what I want to achieve, so now I'm going to work out what I need to do.'" ◢◢◢

becomes about direction and capturing sound in the right way so that it will translate. It's like, 'Let's figure out how do it,' and then you get your head down and you do it!"

Mixing

When it came to mixing, the crew enlisted the skills of one Marta Salogni. "We were all there the whole time," Meredith says. "I think she was quite surprised as normally people just come in at the end, but we were there for weeks, all day every day!" Ross describes Marta as "no ego, just skills." He continues: "Her EQ skills were phenomenal. It's dense music, so you have to make space in it. And she's just masterful. I did some incredibly rudimentary test mixes of stuff and sent it over with all the stems, and within about an hour of working with Marta, the low end is tight, the mix is warm and it's sparkly, you can hear all the parts — and that's just EQ. She's amazing. I don't think we'd even gotten onto conversations about the stereo field or creating space in other ways until a bit later on. It just cleared the path for conversations about creative effects

INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Bob Clearmountain
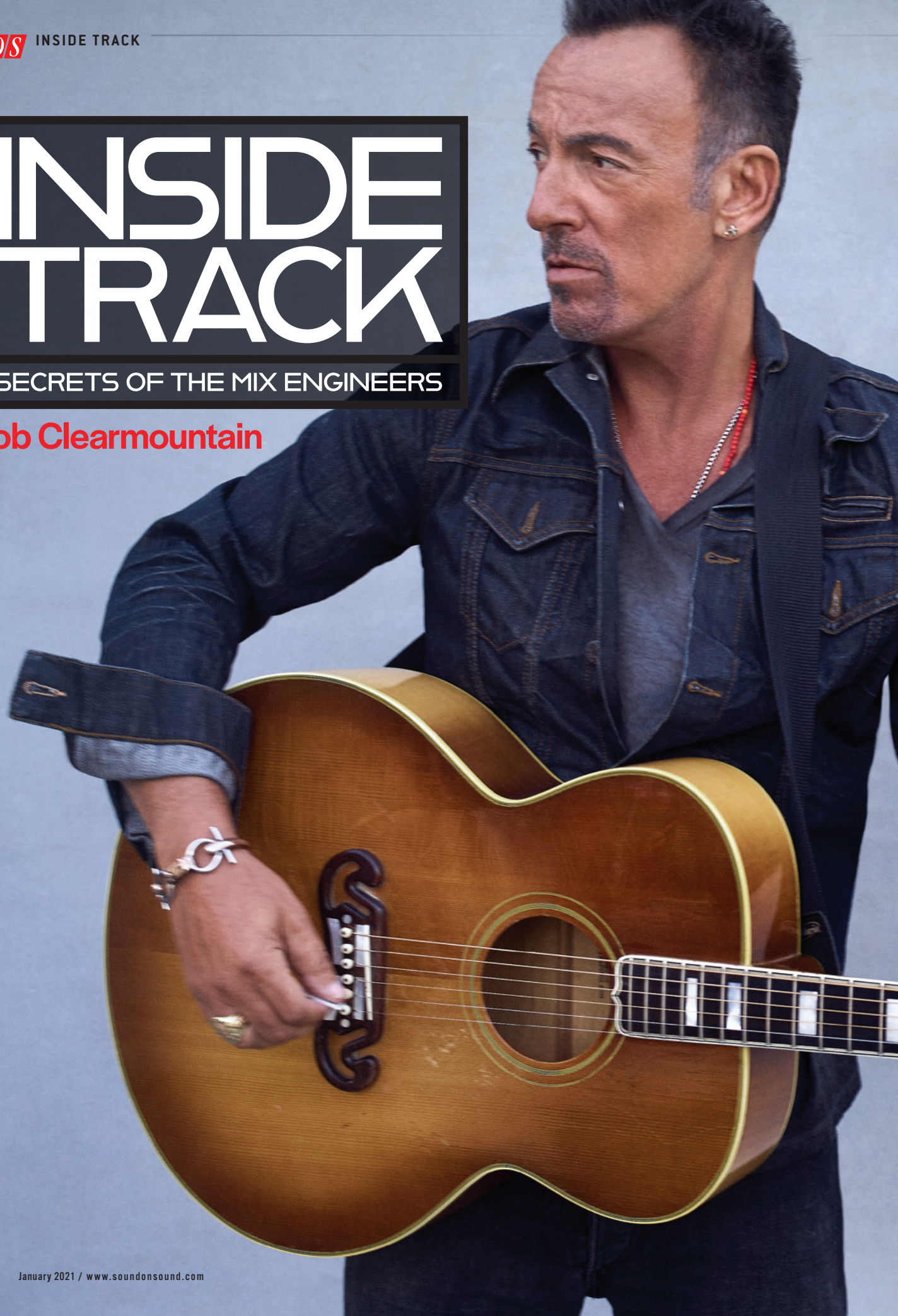
Bruce Springsteen's latest album was an international hit last year, but the console it was mixed on was made a quarter of a century ago. Mixing legend Bob Clearmountain was at the controls.

PAUL TINGEN

"A note to your readers: you figure it out! That's how I learned. Nobody told me what delays or reverbs to use. I would listen to records, and I'd try to work out what they had done, and then I'd try to recreate that, and come up with my own thing. The great part of mixing is figuring out for yourself what works for you, and for the music and the song and the lyrics, and what doesn't. Asking me exactly how I did something is pointless. It really is. I know it is your job to report on this stuff, but dude, it doesn't work this way.

"I learned by doing things," Bob Clearmountain continues. "When I started at Media Sound in the '70s in New York we mostly did jingles, and a lot of the time I was not watching somebody else, but I had to work out what to do myself. So if you're sitting behind a computer, just try stuff. You can do anything you want. But forget about the presets. Presets are for dopey people. With any piece of gear, play with it and listen and learn all the things it can do."

For the legendary Bob Clearmountain, listening is everything, and he prides himself in doing almost everything by ear, and being creative. It shows in many of his working methods, for example the fact that he still mixes on a console, at a time when the vast majority of his colleagues have gone in the box.

"Mixing in the box is like torture to me. I don't understand how people can do that and remain sane. Working with little pop-up menus and windows and so on is just not enjoyable to me. But when I'm mixing at my SSL, it's so much fun, it feels like I'm having a day off! I love the process of mixing on a desk, and the sound also is a big part of that, of course. But most of all, it forces me to trust my ears, rather than my eyes. It's about audio. It's not a visual medium! A few years ago at an AES trade show someone demonstrated some edits in Logic, and then said, 'Let's hear what that looks like.' I started laughing immediately. I thought it was a hysterical thing to say!

"DAWs have many advantages. It's easy, for example, to align several bass drums in Pro Tools. We used to do that on tape: we'd fly stuff out to half-inch tape, and then back in, and we'd use our ears to get them in time. It was more tedious and we couldn't be as precise, but it worked. I don't use Melodyne or Auto-Tune but tune vocals using the Waves SoundShifter plug-in. It's a manual pitch shifter, and I use it the same way I used the Eventide H3000. I just listen and dial it in. I also do this completely by ear. Some things may not be perfect, and that's fine, because you don't want to completely suck the life out of a voice. Today, the vocalists in country music in particular sound like machines. They all sound like the same person, with the soul sucked out."

Star Mixer

Bob Clearmountain inarguably knows a thing or two about mixing records that have soul and life, and about enhancing these qualities rather than stifling them. It was one of the main reasons he managed to become, in the early '80s, the world's first star mixer. The unbelievable amount of classic records Clearmountain has worked on over the decades says it all. They include records by the Rolling Stones, Hall & Oates, Paul McCartney, the Who, David Bowie, Chic, Bryan Adams, Roxy Music, Bruce Springsteen, INXS, Toto, Bon Jovi, Simple Minds, Sheryl Crow, Crowded House, Bryan Ferry, and many more.

Clearly, Clearmountain's biggest achievements were, and continue to be, in the rock arena. And so one wonders how he survives in the current music world, which is dominated by genres like pop, trap, hip-hop/R&B, that are all intrinsically linked to DAW-based working methods, with vocals and programmed music perfectly pitched and timed, instant recalls, and tracks that are compressed and limited for maximum loudness.

Clearmountain stays well away from this particular musical maelstrom, and

'Letter To You'
Written by Bruce Springsteen
Produced by Ron Aniello
& Bruce Springsteen

Photo: Danny Clinch

Bob Clearmountain in the control room at his Mix This! studio.

or massive strings, and I will ask them to just send me a couple of stems of that. But I recently received a session for which they had stemmed the orchestra, and I asked them for the original orchestra tracks, which added up to 600 tracks! But I found it easier to work with that than with the stems, again, because I like the control."

Control Room

The above lists most of Clearmountain's concessions to the modern era. He's also gone as far as digitising his analogue effects approach in an Apogee plug-in called Clearmountain's Domain (about which more below). For the rest his Mix This! Studio, located in his home in Los Angeles, continues to look and function like it did when he built it back in 1994, centred around his SSL and tons of outboard. Before diving into the details of his mix of Springsteen's *Letter To You*, Clearmountain elaborated on his tools of the trade, and what they mean to him today.

"My SSL is from 1994, and one of the last they made. It's a 72-input, 4000+ desk, which has the Bluestone mod, and I later added my own mods, many to do with surround mixing. Surround mixing is a big thing for me. I have done a surround mix for every stereo mix for the last 15 or more years. Sadly, surround isn't particularly important for modern music anymore, so 99 percent of my surround mixes have never been heard.

"My main monitors are the Dynaudio BM15A, which I use for 5.1 as well as stereo, and they're great. I had Yamaha NS10s for a long time, but then Apogee and I were thinking of doing our own speakers based on a newer Yamaha design, which were going to be called

many of the associated working methods. He nonetheless remains not only the most celebrated mixer of all time, but also relevant in 2020, as was illustrated by his mix of Bruce Springsteen's latest studio album, *Letter To You*, which went to number 2 in the US and number 1 in the UK, and in a dozen or more other countries. As Clearmountain-mixed music for the umpteenth time tops charts around the world, it's worth finding out how he works today, and to what extent he's adapted to the latest technology.

"I was one of the first to use Pro Tools, in 1991," comments Clearmountain, "though I'm still not very good at working with it. I must admit, as much as some people get mad at the guys at Pro Tools, this program is really good. They have come up with some amazing stuff. If you don't mind the torture of working purely in Pro Tools, you can do great things. Especially helpful for me is the ability it gives you to fix small things. I do volume automation on my desk, but I'll go into Pro Tools to correct a small detail, like a consonant that jumps too loud or that I'm not hearing. Clip Gain is great for fixing things like that, and you can also use Clip Effects to add a filter or EQ for a brief moment.

"I use some plug-ins, mainly Altiverb, the Apogee ones, the FabFilter EQ, and iZotope to fix stuff, but for the most part I use Pro Tools still as a tape recorder.

In the old days, when we were working with 16- or 24-track tape, the drums were essentially four tracks: bass drum, snare and the entire kit in stereo. All balancing was done while recording, and quickly because you didn't want to keep the band waiting. Now you can have everything separate, including ambient tracks, also for guitars or vocals, which is great when doing surround mixes. You can have endless tracks, and I love the control that gives you in the mix.

"I will for that reason not mix from stems. I'm not interested in that. Once in a while there will be something like a massive backing vocal arrangement

A Star Is Born

It's almost a cliché to point out that Bob Clearmountain pioneered the role of the star mixer, and thereby paved the way for the likes of Andy Wallace, Chris Lord-Alge and Serban Ghenea. But how did he actually get into that position? Certainly taking on the last name Clearmountain (his actual name is Italian) was a masterstroke of branding. But for the rest, did he devise a masterplan, or did it all happen by accident?

"I don't really know how it happened," laughs Clearmountain. "It was weird, because I did not deliberately try to get in that position at all. I realised very early on that mixing was my favourite part of working in the studio. I was producing at the time, which was the early '80s,

and I produced maybe three records a year or something like that, and I mixed nine or 10 records a year. But I hated producing, actually, and I did not think that I was all that good at it.

"To me mixing was far more enjoyable, and more and more people were asking me to mix, even while I was still on staff at The Power Station in New York. My manager said: 'Maybe you should think about going independent, and I can get you a point, for mixing.' Nobody had ever asked for a point for mixing before. Even band members very often did not get points. My manager was a brilliant guy called Dan Crewe, and I thought that what he suggested was crazy, but sure enough people started paying me points! This was around '83-'84."

▬ Bob Clearmountain still uses his SSL 4000 Series console for every mix he does.

» Clearmountain something or other. In the process Yamaha lent us these YSP7 speakers, and I ended up really liking them, so I've since been using them instead of the NS10s. The YSP7s are like the NS10s, but with a smoother top end and the bottom end goes down a bit further, and they're self-powered. But they stopped making them, and we never got round to making the Clearmountain speakers!

"I have two Pro Tools rigs at Mix This! One is my multitrack rig, which runs on the last 12-core tower Apple made, also known as the 'cheese grater'. It connects via three Apogee Symphony I/O's to my SSL. My print rig is a 'trashcan' Mac Pro, and has one Symphony 16-in, 16-out I/O. My print rig is always at 96IHz. Both rigs are locked together with the Apogee Big Ben supplying the multitrack with word clock, and the print rig with video sync."

Outboard Equipment

Clearmountain's outboard includes a lot of classic gear, and he highlights some of his favourites. "I have five Pultec EQP-1A3 tube EQs, which I love. Two of them are the new reissues, which are identical to the old ones. Apogee did a plug-in emulation of the Pultec, and I cannot tell the difference between the plug-in and the outboard, which kind of annoyed me, because I spent a lot of money on these hardware things [*laughs*]. In addition, the Avalon AD2055 EQ has a really nice, warm bottom end, and my BSS DPR-901 dynamic EQ is great for problem vocals.

"As for compressors, I have four Empirical Labs Distressors, which is probably the best-ever modern compressor. They do something wonderful to acoustic guitars, bringing them upfront and making them vibrant and useful. I also have three UREI 1178s, which are the stereo version of the 1176. They are fantastic, and I often use them on vocals and drums. My Neve 33609 stereo compressor is mostly used on bass. I have an SSL compressor which is the same as I have in the desk, and that I tend to use on piano.

"I also have four LA-3A compressors, which are great. The Apogee Opto-3A compressor plug-in is almost identical, so a lot of the time I will simply pop in a bunch of those on backing vocals. I also have a bunch of old dbx 901 de-essers. Other effects units include a couple of Yamaha PCM70s, a Yamaha SPX990, a bunch of Eventides like the H3000 and H3500, though most of them don't work. I still really like the Eventide Ultra Harmoniser DSP4000. I also have an SPL Transient Designer, which I practically never use, and the old MXR flanger/doubler, which is really great."

Clearmountain became famous for his elaborate effect chains, some of which he replicated in the Apogee Clearmountain's Domain plug-in, with a domain being, as Sam Inglis explains in his review of the plug-in, "a blend of treated reverb, delay or ambience that gives each mix its own specific character." (See www.soundonsound.com/reviews/apogee-clearmountains-domain.)

"Over the years I was always experimenting with different things with

---

No Repeats

One pivotal record in Clearmountain's oeuvre that really made his name as a mixer was Roxy Music's 1982 album *Avalon*, a brilliant-sounding piece of work, full of sumptuous delays and reverbs. "People have talked about that record more than about anything else I have done," says Clearmountain, "calling it some sort of breakthrough. It was indeed an amazing record. I'd love to take full credit, but I can't. So much of it was due to the band, the incredible arrangements, and Rhett Davies, the producer. The wonderful arrangements really suited my way of working, so those mixes were quick and not difficult. They took four hours each, and I did two of them a day.

"I guess I was experimenting in the direction of working extensively with delays and reverbs and so on. But also, it suited many of the projects I was working on at the time. In that context,

I have an interesting story about the *Avalon* album. After I finished it, people were just so buzzed about how nice it sounded — big and wide and luxurious, and so on. The next project I mixed was an album for the Australian band the Divinyls, called *Desperate* [*1983*]. I kept the approach of *Avalon* while mixing their album, and it soon became clear that the guys were not happy. After mixing two songs they pulled me aside and said: 'Listen, you have to understand, we are not Roxy Music. We are not big, wide and hi-fi, we are like a little nasty gritty spiky ball, an irritating punk band, so what you are doing isn't working.' I had to start all over. It's often so good to learn from your advice, which would be great advice for a certain soon-to-be-ex-president. You cannot take your approach on one project and apply it to the next project. Each artist's music is unique."

effects and I ended up with signal chains that used my SSL, delays, pitch-shifters, our own convolution reverbs, and other things," elaborates Clearmountain. "I always had my echo patches, and I would patch these into delays and reverbs and I'd modify them a bit for each project. It intrigued me, they were like fun puzzles, with me finding interesting ways of getting particular sounds. Clearmountain's Domain is made up of these different experiments, which are easily customisable by the user. Most of my outboard digital delays have died now, so it was great to recreate them in the Domain, and I now use the plug-in in most of my sessions. The presets in the plug-in were an afterthought. Apogee asked me to do presets of effects similar to those on classic hits I mixed in the past, which I for the most part had to recreate by ear. So they're not all that accurate, to be honest."

Reset & Forget

Because of his motto 'listen and be creative,' Clearmountain doesn't like presets or habitual mixing. It therefore does not come as a surprise that he wipes the slate clean at Mix This! every time he starts a new project. "Everything on the board gets normalised, just set to zero. There are a couple of patches that I usually start with, like there'll be a parallel snare channel, with the parallel having one half of an 1178 and an EQP-1A3, but that's all. I also have sort of a template at the bottom of my Pro Tools sessions, with several aux effects tracks, many of which have my Domain plug-in.

"People tend to send me Pro Tools sessions to mix. I ask for sessions without plug-ins or bussing of any kind, or just a bunch of files. Sometimes it's easier to create a session from files, rather than convert someone else's Pro Tools session, and having to work out what I need to get rid of and what I want to keep. As I said earlier, I don't mix from stems. Of course, I listen to the rough mix, to hear what they did and get the vibe and the feeling they're going for. But I prefer to duplicate what they have done in my own way, especially with regards to delays and reverbs and so on.

"I prefer to start a mix with nothing, because I want to hear the original track sounds before I do anything. I will listen carefully before I add anything. So I'll listen to everything with all the faders up, and then I'll go into individual channels. I'll solo each channel to get familiar with what's there and what contribution each instrument makes and learn about the arrangement. Of course I'll also check if here are any weird noises or rumbles that need taking out.

"The first few passes of the mix is more of a learning process. After that I usually start working on the vocals and guitars. To me the song is mainly about the vocals and if there are a bunch of guitars, I want to know how they balance out and what the arrangement is. Eventually I'll get to the drums and the percussion, and how they relate to each other and the rest of the arrangement. I'll be panning things throughout this process. Stereo and surround mean a lot to me with regards to how everything is arranged.

"Once I have a mix that sounds good with a basic balance, I'll get into

»
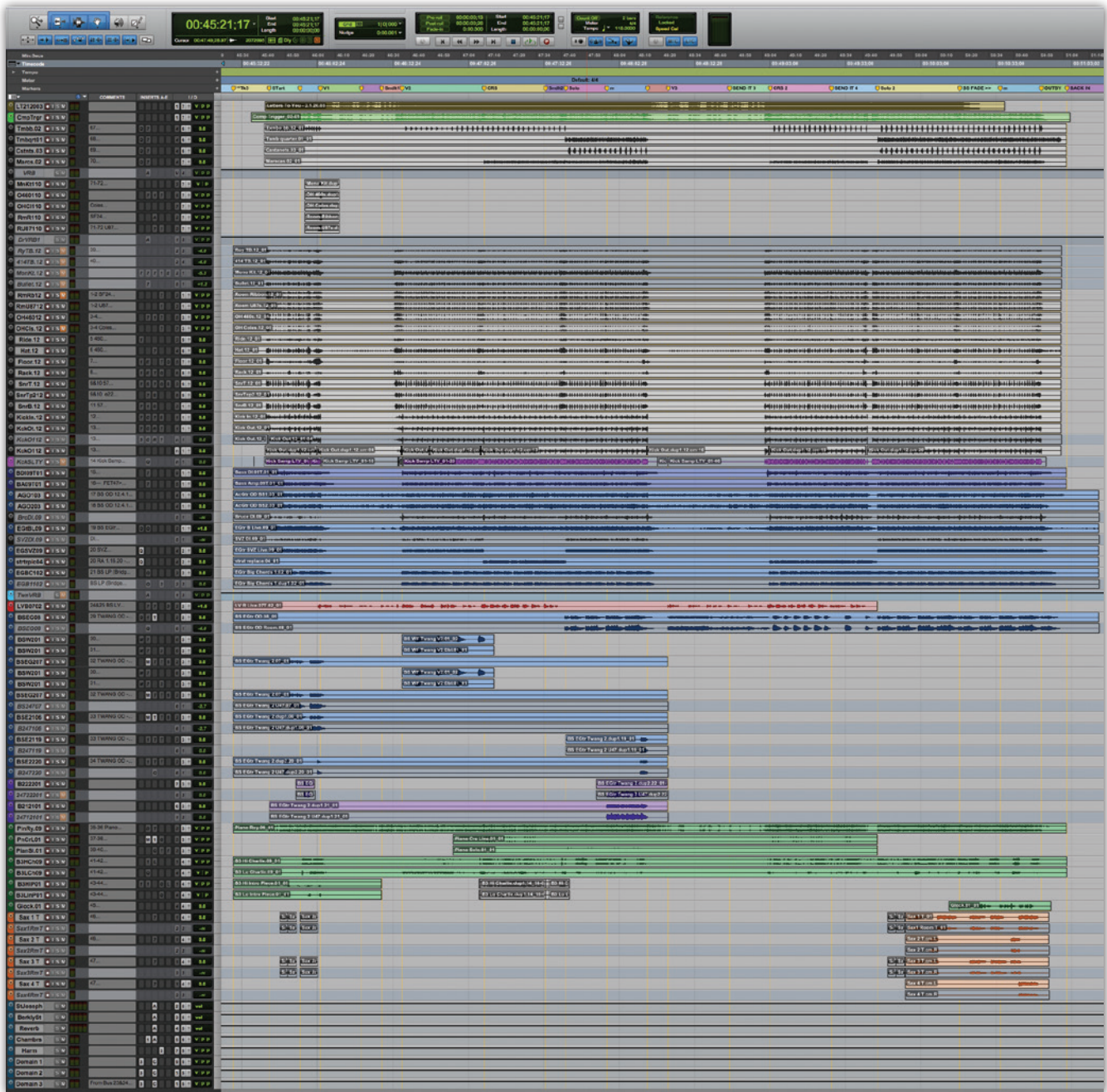
■ The session for 'Letter To You'. As well as the album version, the session also contains the mix used in the *Letter To You* documentary, with an uncompressed mix being used to trigger compression on the stems.

» volume automation. Mixing with the SSL mix automation is by far the best way to mix. It's a natural, wonderful way of mixing. They had listened to a lot of comments from mixers like Frank Filipetti and I over the years, and by the time of the last software update in 1995, they had really nailed it. It's still on floppy disks! It's a shame people don't know about it, because it is so good, especially compared to any current DAW or digital mixer. You can grab anything at any time, and it remembers everything you

do. I don't understand how anyone gets a decent mix in Pro Tools or Logic or any DAW, because they all suck. Ironically, the ancient SSL mix automation is far ahead of any of these systems, and it's fun to use.

"To give one example: you can't playlist your mixes in any DAW. In the SSL you have an unlimited playlist of mixes, and you can edit between them, take the chorus from one mix and the middle eight from another, and so on. It's so much better, it's mind-boggling. Compared to

1995, all these 2020 systems are just pathetic. I asked Avid why they don't make it possible to playlist your mixes, and they said: 'Do you know how much information that is?' I said that all they really had to do is volume — in the SSL most of the rest of the mix is static. But they said, 'it's either all or nothing.'"

'Letter To You'

The session for the title track and first single of Springsteen's 20th studio album, *Letter To You*, contains about 60 active audio tracks, and eight aux effect tracks at the bottom. A small number »

of audio tracks and three aux effect tracks in the session are de-activated. From top to bottom the session contains a rough mix and a Comp Trigger track (explained below), drums and percussion in black, bass and guitars in blue, lead vocals in red, more guitars in blue and purple, piano and Hammond B3 in green, saxophones in orange, and the aforementioned aux effect tracks in blue. The aux effect tracks consist of four tracks with Altiverb (St Joseph, Berkley Street, Reverb and Chambers), a Harmonizer track, and three tracks with the Clearmountain's Domain plug-in.

"This session was weird," explains Clearmountain, "because there's also the *Letter To You* documentary, and some of the sessions were more for the film mix. So there's a rough mix track at the top, and the Comp Trigger track underneath is an uncompressed stereo mix, which is there for making stems for the movie. We used that to trigger the SSL stereo bus compressor, so every stem gets the same compression.

"The greyed-out plug-ins and tracks in the session are all from the Pro Tools session I was sent. The first thing I do

is get rid of the plug-ins. I tell people: 'If there's a specific sound you want, like distorted guitars, then just print it that way.' I will do EQ on the SSL, and the compression with my outboard, and many of the other effects with my aux tracks.

"The snare drum parallel here indeed has the UREI 1178 and an EQP-1A3 — the latter is actually broken, but I like it. I boost some 5kHz, really broad, to add overall brightness, and I also push 100 cycles,

> ## "Mixing in the box is like torture to me. I don't understand how people can do that and remain sane."

just to warm it up a bit. The main snare track will just have EQ on the desk. I also often add a bass drum sample to give some more solidity. The purple track in the session is their bass drum sample. In general I will brighten overheads. They had pairs of rooms mics like Neumann U87s, Coles ribbons, and I would have listened to how they sounded, and balanced them and if necessary added some of my own reverb or ambience.

"The reverbs I use depend on the song. Ballads tend to have longer reverbs. Sometimes only using the room track

that's in the session works great. If I add reverbs, it'll often be from Altiverb, like Berkeley Street (the original name of the Apogee Studio), or, very rarely, St Joseph Church, which is too big for drums, and better suited to choirs and strings. Occasionally I use Altiverb's Castle de Haar. It works well on things you normally use a plate reverb on, but I think it sounds nicer than plate. It's warmer sounding and not quite as splashy. The Chambers auxes are my live rooms in Mix This! They now are used as storage spaces, so I have to use the Altiverb versions.

"With the bass, many people record the DI and the amp, and the latter will be ever so slightly delayed because of the circuitry the sound is going through. So if necessary I'll align them the waveforms so they're perfectly in phase. This tightens up the bass sound. I usually push both bass channels through the Neve 33609 compressor and the AD2055 for a bit more warmth in the bottom end.

"I mentioned that I love using Distressors on acoustic guitars, so I also did that here. Many compressors sound like they fight the particular dynamics of many acoustic guitars, whereas Distressors don't. I would also have added some reverb to the acoustic guitars. In general I try to use a combination of similar reverbs on many of the instruments and the vocals. I don't like this thing where each instrument has its own specific reverb, and it sounds like they're all in different places. When I hear a record like that, it really annoys me, because I wonder: 'Where is this place?' It doesn't make sense to my brain.

"On electric guitars I will often use some slap delays from Domain 1, maybe 90ms on one side and 125ms on the other side, just to give it a bit of aliveness, and then mix some reverb in with that. On the Pro Tools inserts of some of the guitars and keyboards I use the FabFilter Pro-Q 2 EQ, the Apogee compressor, and the Soundtoys Tremolator. I don't really like to talk about exactly what I do in specific songs. It totally depends on the song. I prefer it to remain a mystery. Once again, listen, and then try things yourself!

Vocals & Mix Bus

"With regards to the vocals, I listen to them and try to get a feeling for what the song is


▬ Bob Clearmountain's collection of outboard processors include a number of UREI 1178s, Empirical Labs Distressors, and various Eventide multi-effects units.

Total Recall

While many mixers would love to still be working on a desk, they don't for the simple reason that they need the capacity for instant recall that a DAW provides. Clearmountain, however, says it's not an issue for him. "Recalls are simple. Years ago I came out with this thing called Session Tools, that Apogee used to sell, which includes templates for invoicing and work orders and labels for tapes and CDs and so on. I have since modified it pretty heavily for my own usage. We have a whole library of pictures of the outboard gear that appear in a FileMaker layout. I print out the ones I need, and take a pencil and then mark the settings. It takes a few minutes, but it comes back perfectly. When possible, outboard gear is set to unity gain and most of the delays and 'verbs are in the box and saved with the session. It takes half-an-hour to set up the desk because you have to do it manually, but I think of that as a sort of Zen time of contemplation. The computer remembers every knob, it comes up on a graphic display which is like graphics from the '70s. It is one step above the *Pong* video game, but it works. I do recalls all the time. It's even easier and quicker when I have the budget to pay my assistant to help, who works part-time."
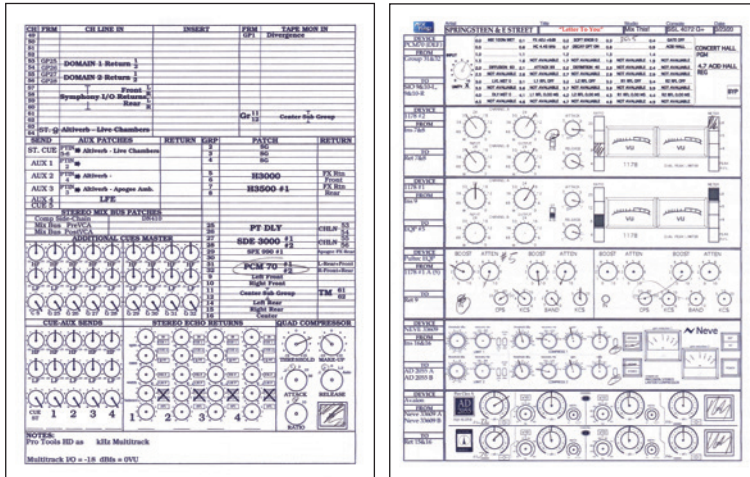
Outboard and console recall sheets from the mix of 'Letter To You'.

about, and I then try to figure out what the narrative of the song is, what's in the head of the singer or the main character in the story, and to put myself in that place. So I try to get the feeling, and I then try a bunch of different effects to see what works the best. Sometimes it's nothing. The vocals in the Crowded House song 'It's Only Natural' are completely dry, because, well, it's only natural! With a dry vocal you feel really close to the singer, who is right in your face, and I love that. You can have a dry verse and then some space in the chorus to expand and create a contrast. With Bruce's voice I tend to go for a little bit of slap delay and some ambience. I think I had the Berkeley Street reverb on him."

And finally, while it's commonplace for mixers these days to have extensive signal chains on the master bus, partly in the service of maximising loudness, it's another area where Clearmountain just absolutely will not play ball. "I print my mix straight from the desk. At most I'll have a little basic compression from the SSL desk compressor on the mix. I compress mixes of live tracks more, because it adds to the excitement. I get within a couple of dB of going over, because you want to use as many bits as possible, but that's it.

"I don't go along with the whole loudness thing at all. Everything should have a place in the mix, but your overall mix doesn't have to be super-loud, because mastering and all the streaming services deal with that. My theory is that the whole loudness thing came about because at A&R meetings each A&R person wanted the tracks they brought in to be as loud as possible compared to all the other stuff. So they'd have an engineer send a final mix through a Waves L2 or something, and boost the level beyond comprehension. But it sounds like crap. It's distorted and you lose all dynamics. I enjoy dynamics in a record. I don't care about things being very loud. I want my mixes to sound like music!" ◢◢◢

[**SAMPLE LIBRARIES**]

Orchestral Tools
Modus Jeff Russo
SINE Player
★★★★★

Drawing on composer Jeff Russo's work for *Star Trek: Discovery*, Modus contains 21 ensembles recorded at Teldex scoring stage from up to six mic positions. Like Orchestral Tools' Time Macro and Time Micro collections, the library (29GB installed) avoids the hacking strings, blasting brass and pounding percussion of conventional cinematic titles and focuses on subtle, atmospheric evolving textures and imaginative performance styles.

Mr Russo has conjured up some unusual, ear-catching combinations: alto flutes and violas (a beautiful, lyrical woodwind/strings texture), piano and violas (great for midrange ostinatos), and the sparkling celestial chimes of unison vibraphone and glockenspiel, described by Russo as "the sound of Trek". Ramming the point home, the library also features a Klingon choir intoning guttural syllables in their own language, which is frankly what the sample world has been waiting for.

In addition to these otherworldly timbres are human male and female 12-piece choirs, large string, woodwind, trumpet and horns sections and a low brass combo of bass trombones and cimbassos, augmented by tuned percussion ensembles and lush cymbal effects. I enjoyed the low strings' big, brooding, ominous sound, the calm, tranquil tone of the six clarinets, the eight horn players' broad, warm tone and the choirs' quiet 'mms' and 'aahs'. Also featured are a pair of Japanese shakuhachis (another Trekkie sonority) whose breathy legatos add a sinuous world-music aura.

All good stuff, but Modus's heart really lies in its dramatic performance styles. 'Transitions' are ready-made note movements with a built-in articulation change (for example, normal to tremolo, or vice versa) with an initial short swell leading to a held note which then fades down. You can set the target to one or two semitones up or down, or leave it at unison. Thus you can trigger uplifting melodic and chordal movements played by high strings, low flutes or octave bassoons, shuddering low strings descents, woodwind and brass flutter-tongue outbursts and choral intervals shifting from 'aah' to 'ooh' vowels, all at the touch of a finger.

'Tension performer' artics introduce independently played additive harmonies when you raise the mod wheel. There are four types: 'overtones' correspond to the natural harmonic series, 'fifths' stacks up quiet perfect fifth intervals, 'cluster' adds four or five semitones to
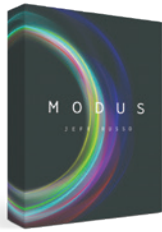
cacophonous effect, while 'atonal' ticks the horror and suspense boxes by introducing a series of increasingly dissonant notes. The results are revelatory, inspiring and occasionally scary — the female singers' quiet clusters are not the sort of thing you want to hear when visiting a deserted graveyard at midnight on All Hallows' Eve.

On a more cheery note, you can now buy each of the 21 ensembles separately for as little as €10, then download their individual mic positions when you need them. Hats off to OT for championing this inclusive, forward-looking approach. *Dave Stewart*
€349
www.orchestraltools.com

Murst Instruments
Glass
Kontakt Instrument
★★★★

Glass is an instrument designed to generate long, evolving textures combining both pitched and unpitched looped samples derived from synths, strings, woodwind, field recordings and, in some cases, subjected to granular processing. Evolving pads is what this synth does best and it does it in a very endearing way via its almost childishly simple interface. Requiring the full version of Kontakt 6.3.1 or later, Glass is presented as four different sample-based instruments, and each instrument is made up of four sound sources created from permutations of 16 samples per instrument, each source having individually adjustable AR envelope, level, pan, ambience reverb, filtering, phase effect and a 'smearing' control. Kontakt's standard 'right click to learn' CC message is supported by Glass as well as an MPE engine to take advantage of controllers such as the Roli Seaboard
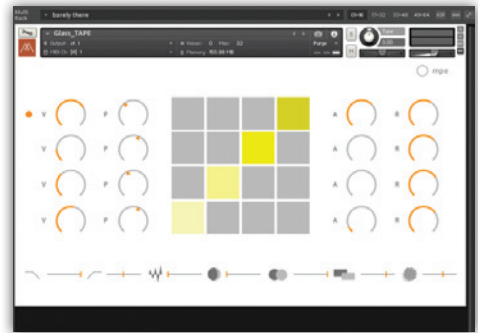
All four instruments follow the same format other than showing different coloured squares loaded with different samples, and these instruments are designated Swell, Resonance, Tape and Glitch. Swell hints at acoustic instruments in the context of long and evolving sounds while Resonance leans more towards vibrating resonant items such as drums, pipe or strings. Tape apparently uses samples that were processed via a Roland Space Echo while Glitch throws away the rule book by mangling both analogue and digital waveforms to produce pulsing soundscapes. Because Glass uses long looping phrases, the combination of sounds evolves in a very organic kind of way.

To my ears all four of the instruments are capable of producing wonderfully evocative evolving pads or underscores that work well either on their own or layered with other sounds.

Though simple in concept, Glass offers a lot of potential both for creating underscores and pads and for layering with more conventional synth sounds. The way in which sounds evolve is very appealing and because of the independent resonant filters and LFO filter modulation afforded to each of the four layers, it is possible to coax much more variety out of this seemingly simple format than you might think. *Paul White*
£79
www.murstinstruments.com

Cinematique Instruments
Rytmik
Kontakt Instrument
★★★★★

Cinematique Instruments always seem to do something slightly leftfield with their virtual instruments and Rytmik, while ostensibly a sample-based drum and percussion library aimed at media composers and electronic

music producers, is no exception. At its heart is a collection of over 200 unique drum, percussion and effects sounds, delivered as a Kontakt-based instrument. The underlying instruments go from massive kicks and booms all the way to intricate percussion sounds and the quality is uniformly high. A collection of 350 presets provide instant gratification but you can also build your own by combining up to 24 of the underlying sounds, sequence each sound with its own 32-step sequencer, trigger any or all of the 24 sounds via MIDI, and apply a range of effects and real-time sound tweaking via the mod wheel.

The main controls are spread across four tabs: Main, Mixer, Blocks and Shape. Main provides access to the preset browser with filter options to narrow down your search. A well as conventional mixing duties, the Mixer is where you can browse and load individual sounds. The 24 sound slots feed four group channels (including the option to load groups of instruments) and you also get reverb and delay send effects and a global compressor.

Block provides the step-based sequencer with a velocity lane for each of the 24 instrument slots. Clicking the small 'e' button on the left of any lane expands the editing environment for that lane. You can trigger playback of the whole preset here or via the C4 key. Alternatively, you can trigger the sequencer for combinations of sounds in the notes above C4 or play individual sounds in real-time via a keyrange below C4. These keys are all colour-coded on Kontakt's virtual MIDI keyboard. There are some cool options here including shifting beat patterns along a lane and live recording of patterns, making this a very creative working environment. Finally, the Shape page provides options to configure how velocity, volume, high- and low-pass filters, reverb, lo-fi and distortion effects are modulated by the mod wheel. This is simple to configure but very effective in use and a small number of 'Shape' presets are instantly available as the base of the page to get you started.

The UI is classic Cinematique Instruments (cool but slightly quirky!) and, while it remains easy enough to find your way around, there were a few graphical tweaks that might makes things easier for the user. For example, in the Mixer, the central circle lights up when you trigger the Play button but (for me at least), not when triggering individual loops from my MIDI keyboard. Equally, in the Blocks page, it would be useful if the step sequencer showed a playhead on individual lanes (instead of just when the full pattern is in playback) so you can more easily see which instrument you are triggering.

These are minor wishlist comments though because, in use, Rytmik sounds excellent. It's an impressive sonic arsenal presented in an interesting and creative interface and will undoubtedly appeal to modern media composers or electronic producers. *John Walden*
€170
www.cinematique-instruments.com
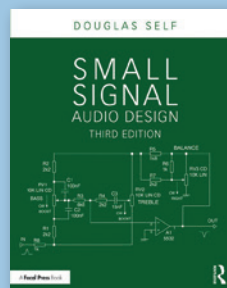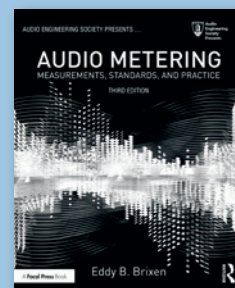
Q How do I connect mic preamps to patchbays for line-level processing?

I enjoyed Hugh Robjohns' excellent article 'Patchbays For The Modern Studio' [in SOS *December 2020*] but one point he made left me confused: he said that you shouldn't put mic preamp inputs on the main patchbay but, instead, give them their own XLR inputs. I get that this removes the risk of phantom power being routed to the wrong place — but I often use mic preamps as line amps to add transformer coloration and the mic and line inputs use the same physical connector. I also have a 500-series rack, and sometimes swap the modules around so slots might or might not host preamps.

I'd prefer to have access to all these 'line' devices from my main patchbay. What approach would you recommend as the best low-risk strategy? Maybe just have a separate XLR panel and use male and female XLR-to-TRS patch cords to link to the main panel, so I have to think about it?
Andy Sullivan

Technical Editor Hugh Robjohns:
You can certainly use TRS-XLR adapter cables to get line sources into mic preamp inputs, and often that's the most practical option. For 500-series racks, if

they're likely to host any mic preamps, then I'd suggest connecting the inputs to the mic patchbay too. Alternatively, consider a dedicated 500-series chassis for line-level processors and another for mic preamps.

However, I'm not a fan of draping long patchcords around the place between bays, and especially not on a regular basis. So if patching line sources into preamps is something that's likely to be a common requirement, I prefer to build in some dedicated tie-lines to go from your main line-level patchbay across to the mic preamp patchbay behind the scenes.

Typically, I'd have a couple of sockets on the main patchbay labelled 'Mic Patch Tie-line 1/2' or similar, and on the mic patchbay I'd have a corresponding number of male XLRs labelled 'Main Patch Tie-line 1/2'. With this arrangement on your main patchbay you can patch your line output into a tie-line using a standard patch cord, and then using an XLR-XLR patch cable, you can link the tie-line output into the preamp input on your mic patchbay.

Obviously, this isn't a foolproof solution; you could still accidentally leave phantom power on the preamp which would find its way back across to the line-level source. However, I find the act of patching up tie-lines forces me to think about what I'm doing and that reminds me to check the phantom is off before I complete the patch!

For a guaranteed idiot-proof system, though, you could install suitably large DC blocking capacitors or an isolating transformer in the tie-line wiring between the two patch bays to ensure phantom power can't make its way between the patchbays. The capacitor should be chosen for the device with the lowest output impedance: the lower the output impedance, the bigger the capacitor needs to be. In practice, a capacitor of 10uF or larger will do the trick.

Q What's causing distortion in my in-ear monitors?

My church is using Shure SE-215CL in-ear monitoring with PreSonus HP2 personal amplifiers, and the sound is distorted at higher volumes. I have checked the gain staging but I'm thinking the headphones and amp aren't a good combo. Please tell me if the headphones and amp we have should work together — and if not what amp or what headphones we should get to fix the issue. I've been trying to figure it out but it's so confusing!
SOS Forum Post

Hugh Robjohns:
I suspect the problem here is not your choice of amp and headphones, but of gain structuring. The specs for the HP2 are confusing, in that they use random amplitude references in quoting a maximum input level of +4dBV and an input limit of +5dBm. But the bottom line is that the maximum input level to the HP2 appears to be around +5dBu, which isn't a lot when most consoles will be able to put out +20dBu or more. So getting the gain structuring right from your console will be critically important to avoid overloading the amp inputs. On an analogue desk, you'd need to keep the sends below 0VU, and on a digital desk, below -20dBFS.

As for IEM/amp compatibility, the HP2's output power is quoted as 50mW per channel with a 20Ω load. That's fine, as your Shures are close to that at 17Ω, which is also comfortably above the 12Ω minimum impedance specified for the amp. The sensitivity of the SE-215CL is pretty high at 107dB/mW, which means the HP2 amp should be able to generate around 123dB SPL into the performers' ears. That ought to be more than loud enough, seeing as it's the nominal
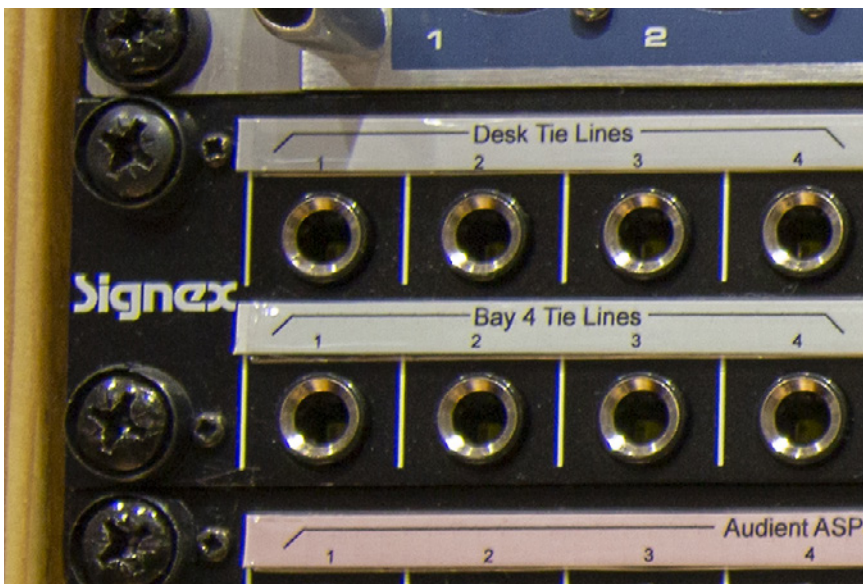
When working with more than one patchbay, for example one for mic inputs and another for line sources, it can be a good idea to set up 'tie-lines' between them.

SOS TUTORIALS

SOS

Get Started with Sound On Sound Tutorials today!

1,151
COURSES IN LIBRARY

2,236 hrs
OF EXPERT TRAINING

Login

Sign Up

Start learning with **Sound On Sound** today.

Stream courses to any device!

Sound On Sound have partnered with NonLinear Educating to deliver a high quality
training and learning experience using their extensive range of popular video courses.
Buy just the courses you want or buy an Unlimited Access Library Pass
and watch any course video at any time for a low monthly cost.

www.soundonsound.com/**tutorials**

If the amp for your in-ear monitors is distorting, chances are it's simply that it's being fed too hot a level from the mixer!

threshold of pain! If users are complaining that they're not loud enough, it might be worthwhile checking their earpieces for wax blockages, and to ensure they're fitting the earpieces correctly.

Q Can I reduce latency when using Dante Via and Cubase?

My little basement studio consists of one room for recording loud stuff and another as my control room, with an 18m cable run between them. Typically, I use up to six microphones in the loud room, and need maybe three cue returns for headphones. Analogue multicore wasn't an option, as the large holes in the walls this required weren't acceptable, and neither optical ADAT nor USB could cope with the distance. So that left Ethernet.

Consequently, I rigged a system comprising an RME Babyface interface with a fast, desktop PC in the recording room to handle the mics and headphone feeds. In the control room I had a Gigabit switch feeding the recording PC with Cubase 10.5 and an RME UC interface for monitoring. Dante Via software connected the two machines, with Dante Controller for routing. Everything clocked to 48kHz, and it all works... except that the latency is killing me! According to Cubase, it is 30ms, and I have to manually move the files on the timeline to sync playback and recording.

Audinate's Dante Via software does allow you to send audio over a network between two computers, but without any Dante hardware the latency involved will be too much for applications in typical music recording studios. However, there are affordable zero-latency analogue options for connecting a live room to a control room that will make use of the same Cat-X network cables.

I cannot be the only person connecting two rooms with Dante, so is there a smarter way to use Dante Via for this? Or is there a smarter, better way to connect these two rooms?
SOS Forum Post

Hugh Robjohns:
The fundamental problem that you're struggling with here is that you have software Dante interfaces on both computers. Dante was initially designed as a hardware interface, and professional Dante-equipped consoles, preamps, stageboxes and so on, for which low latency is absolutely critical, use dedicated hardware network interface cards.

The Dante software interfaces are designed for use in situations where latency is not critical, such as a laptop eavesdropping on the network and recording everything for use afterwards. With the hardware Dante network interfaces, the console firmware interacts directly with the physical hardware, so data is transferred very quickly. However, software Dante interfaces have to negotiate through the host computer's operating system, and its own network handling software, and that all takes time. Consequently, the typical latency for Dante Via is around 10-15 ms. In your case, as you're running two instances, the total end-to-end latency is going to be around 20-30ms — which is exactly what you've found.

If you want the typical low-latency (1-3 ms) that Dante is rightly praised for, you need to use the dedicated hardware interfaces, I'm afraid. In your case that would typically be a Dante network card in the control room PC and a hardware mic pre with Dante interface in the 'loud room'. And that won't be a cheap option!

However, what would be a cheap option — and still make use of the physical network cabling you've already put in place — would be one of the multichannel passive analogue interfaces designed to connect to standard Ethernet cables. Lots of companies now make hardware boxes for this, and they simply use the four twisted pairs of a standard Ethernet cable to carry four balanced analogue signals. Obviously, this is a point-to-point analogue connection (you can't route it via your Gigabit switch) and you need to use screened Cat-X cables (ideally individually screened Cat-7). But it is relatively cheap, and you get four completely latency-free analogue audio channels per cable.
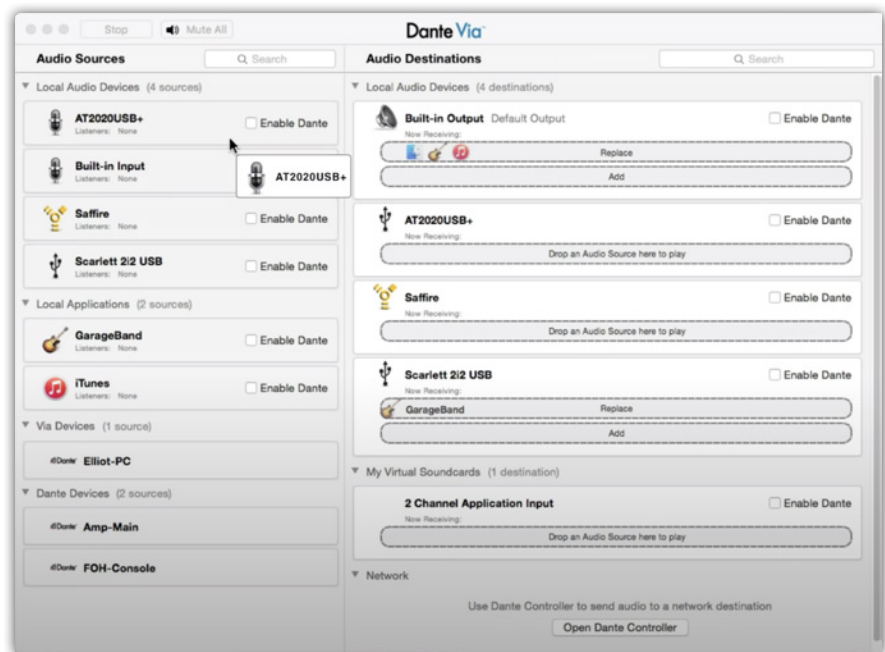
David Carson (Regional Sales Associate): david.carson@soundonsound.com

Martin Christie's ELECTRONIC MUSIC ♥ ♥ #OPEN MIc NIGHT!

SIMON HORROCKS

As an electronic music maker, I've felt the heat when playing at open mic nights where unplugged rock, pop and folk is the norm. Firstly, I'll spend more of my allocated slot setting up my synths and cables than performing, and secondly, my set leaves the audience clueless as to what I just performed and which bits I did live.

After I'm done, I get the systematic wry smile and a mispronunciation of my name from the event host. Cajon chap smiles politely, but I think he's just relieved to give his hands a rest. Guitar guy is oblivious to everything, although he can't stop staring at my Big Muff pedal. Perhaps my synthwave-cum-chillstep piece was just a bit too 'out there'?

So, if these regular open mic nights aren't quite right for the emerging army of knob-twiddlers, loopers and beatmakers, where do we go to perform? Luckily, some innovative new communities are emerging for the growing number of DIY circuit-benders and modular geeks.

Manchester-based musician Martin Christie is responsible for one such platform. Over the last few years he has established a series of live Electronic Music Open Mic (EMOM) nights, which started off as a showcase for his own poetry and electronic folk performances, but has turned into an international, multi-networking, 'anything goes' forum for electronic musicians of all abilities and genres to share new material.

"As much as we love the diversity of regular open mics, electronic musicians need a space where electronic music is understood, and the setup times are more sympathetic," says Martin. "We've challenged venues to create a forum for meeting like-minded synth-nerds and audiophiles."

We are seeing a huge surge of DIY musicians, taking what they learned on Ableton, then throwing away the laptop and plugging things in. Artists like Inwards, Kelpe and Nathan Fake mainly originate from DJ culture, but make live beats, with new sub-genres emerging from ambient, noise, drum & bass, synthpop, experimental and hip-hop, using the implausible hotchpotch of old and new technologies from '80s grooveboxes to crazy suitcases full of wires.

Podcasts like Podular Modcasts, Homebrew Electronica, and Not 97 are great places to explore these under-the-radar sounds. Electronic music-making events have also cropped up over the last few years, such as *SOS*'s own SynthFest, Seventh Wave and Machina Bristronica in the UK, as well as Synth Expo in the US. But for everything live, 2020 has obviously brought new challenges.

Martin Christie continues: "Lockdown hasn't stopped us at all. Within weeks, collaborators and musicians were setting up virtual EMOM live streams which, in turn, created innovative visual performances, even using VR. During and after the streams, performers go online to share links, tracks, get feedback or tips on marketing or distribution. We were also some of the first people to put on socially distanced events too, which has been a real success."

Lockdown has had some positives too. There's time to do more purchasing of gear, writing, recording, rehearsing or trying new things, as well as working on marketing and social media. But clearly, the desire and objective is to get back to the physical, live performance.

Let's just hope that when things are back to normal, more venue owners are open to these new ways of making music, and embrace this incredibly important movement. ▰▰▰